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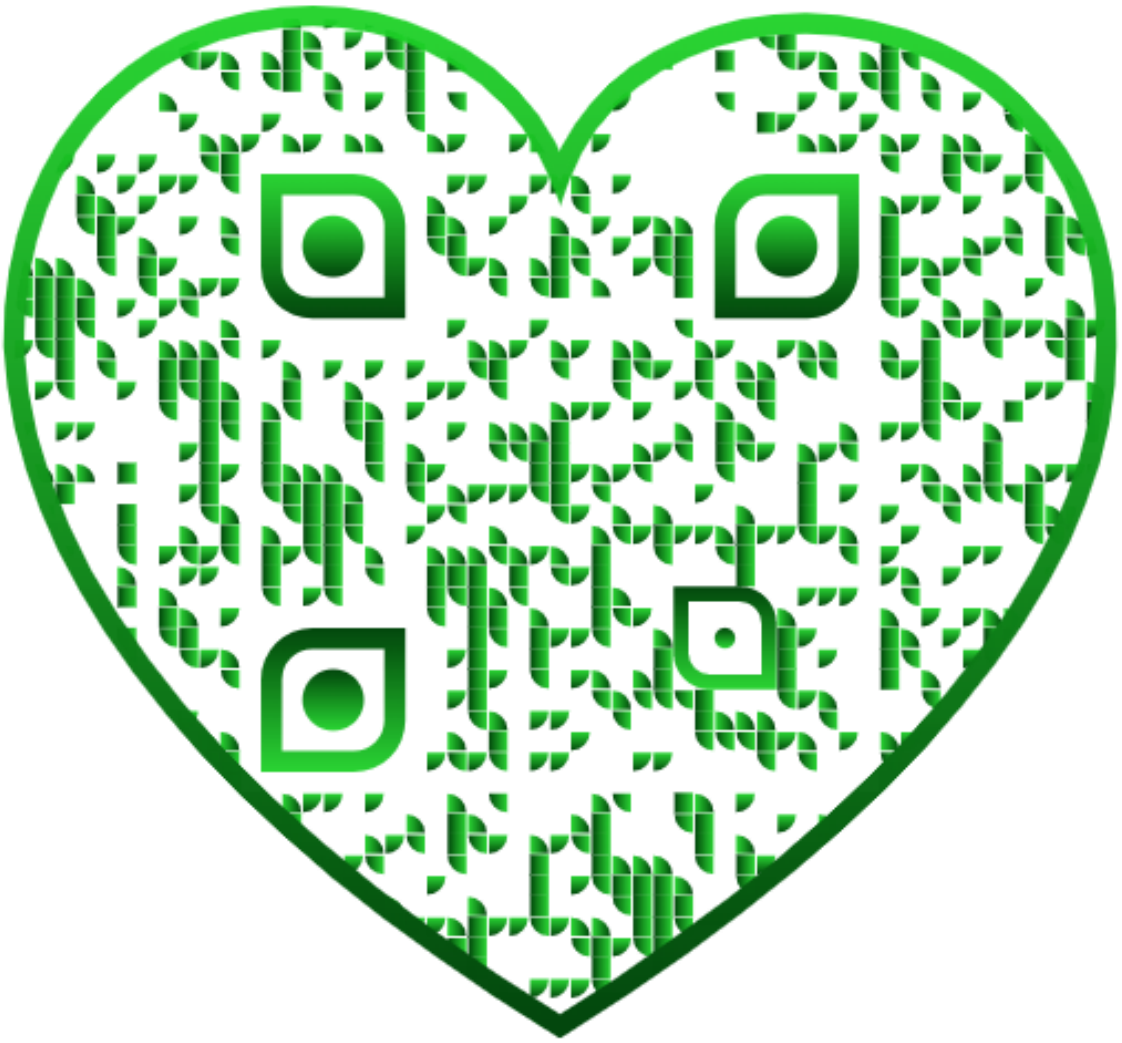

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LOST IN SPACE

Spatial audio. 3D music. 360-degree sound. Call it what you will, immersive audio is a hot topic. And while researching this month's cover feature, I've had lots of conversations with people on the front line. In order of enthusiasm, they rank roughly as follows:

(**1**) Apple. (**2**) Loudspeaker manufacturers. (**3**) Engineers who already have Atmos rooms and are earning good money doing immersive mixes.

Others, however, harbour reservations. Notably: (**1**) People who think Apple are implementing immersive audio all wrong, and would be delighted to explain why their own system is much better if only anyone would listen. (**2**) People who think it's all a gimmick. (**3**) Engineers who don't yet have Atmos rooms and are not earning as much money as they once were.

The most important people in all this, though, are the audience, and their verdict is still awaited. Have Apple and Dolby done enough to entrench Atmos as the new default format for music consumption? Will there be a watershed album that brilliantly showcases immersive production and carries the music world with it? Only time will tell, but it's likely that the pressure to create immersive music will be relentless for the next couple of years. That pressure will be felt by mix engineers, with labels now expecting fully fledged Atmos mixes for no extra fee, even though many of us simply can't afford to turn our mix room into a 7.1.2 environment.

It also raises interesting questions about music itself. At present, immersive music is generated in post-production, from projects that are otherwise finished. Apple's implementation of Atmos within Logic 10.7 looks set to change this. Not only does it democratise the technology, it encourages music-makers to think immersively from the start of the production process. If that takes off, spatial audio really will be a revolution.

Although Apple have lowered the cost bar dramatically, getting into immersive production is still a major commitment in terms of time and effort. Like all big decisions, it's best made from an informed standpoint — which brings me back to this month's cover feature. Before you can inform others you have to inform yourself, and when I tried do this, I found it very frustrating. Consumer sources gave me reams of breathless hype. Technical manuals and standards documents produced an avalanche of impenetrable jargon. What I couldn't find anywhere was a simple, comprehensive and accurate summary, written with musicians and producers in mind. So that's what I set out to write, and I hope you'll come away from this issue with a clearer idea of what immersive audio could mean for you.

Sam Inglis
Editor In Chief

"Although Apple have lowered the cost bar dramatically, getting into immersive production is still a major commitment in terms of time and effort."

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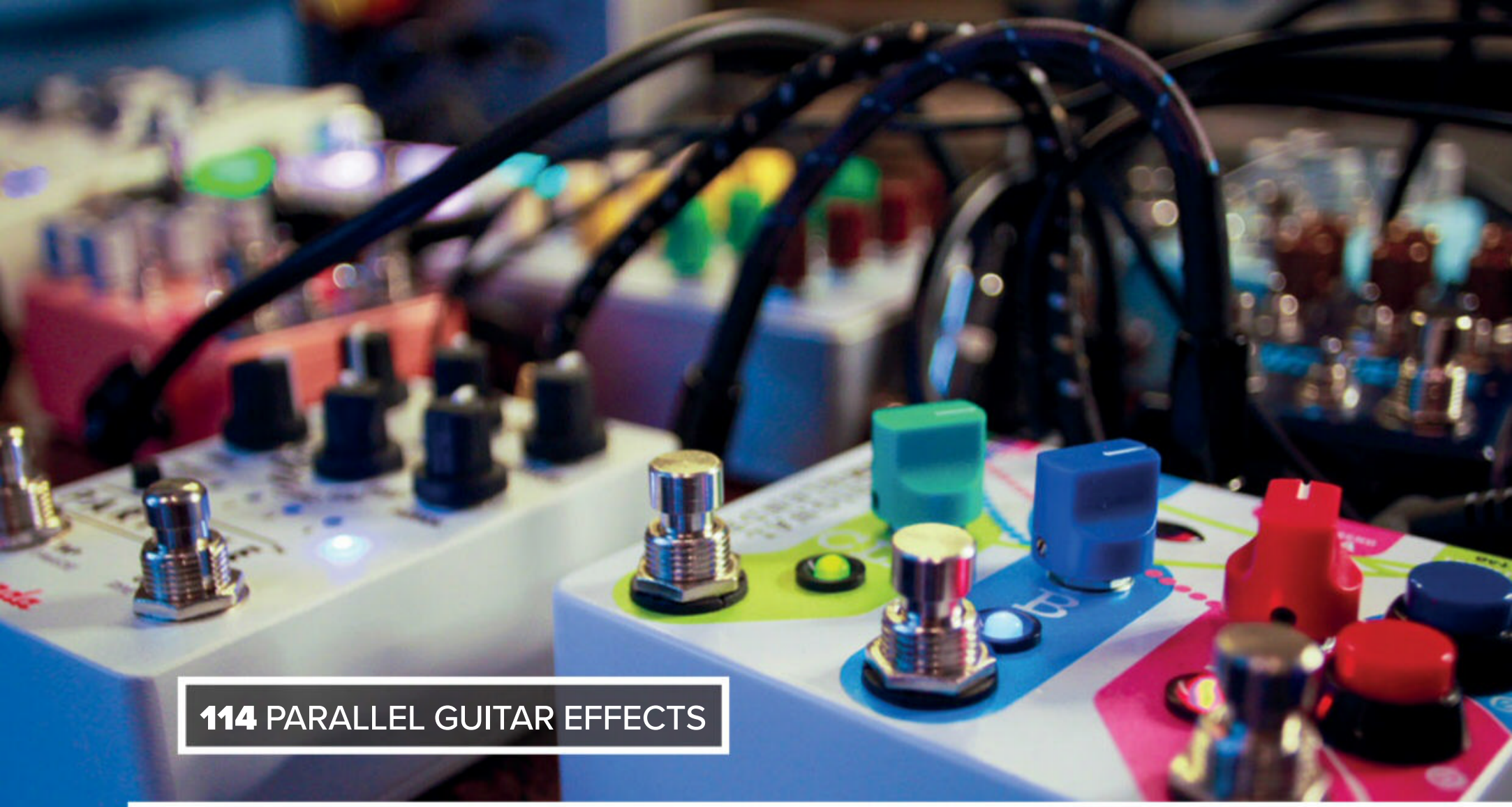

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Black Rooster Omnitec-67A

Analogue-modelling Plug-in

The 67A adds some useful features to a nice Altec valve preamp emulation.

NEIL ROGERS

Black Rooster Audio were founded in 2016 by Andre Kirtchner and, from their base in Wuppertal, Germany, have released a steady stream of plug-in emulations of vintage compressors, EQs, preamps and effects processors. Their latest release, Omitec-67A, sees them visit a classic valve preamp design from the 1960s that was used on some of the most celebrated music of that era. As with much vintage gear, it's not the functionality of the design that's coveted but rather the more characterful side effects they can impart: the rasping, saturated brass sound on many of the classic Motown-era records is just one example of how engineers used gear like the Altec 1567A for more than just raising the signal level of a microphone.

The original Altec 1567A has very little going on from a controls point of view, offering just gain (a whopping 97dB, mind!) and a simple two-band EQ. Black Rooster's homage to this design aims to capture the essence of the original, whilst also offering some additional features including an extra 'mid' EQ band, three different tube options and a 'bias' control. There's also a mix control for parallel processing, but no obvious option for linking the input and output controls which, for a dedicated saturation device, feels like an obvious omission.

Sat Tests

Bass guitar seemed like a great place to start and I could hear the subtle saturation effect straight away with the default gain setting, which had the effect of making the more audible midrange frequencies feel more 'alive'. There's the option of switching from a line to microphone input, which mimics the change in impedance — this is a bit like increasing the input level to drive the 'circuitry' that bit harder, though of course impedance also varies with frequency. When engaging this input control, the internal signal flow of the plug-in compensates for any increase in level at the output stage, so this is a great way of getting a quick feel for how much character you want to impart. It also goes some way to addressing that minor grumble about not being able to link the input and output controls.

On drums, I liked how putting an instance of the 67A on my drum bus helped to tie the sound of a live drum kit together into more of a cohesive sound, especially, how it pulled the top-end of cymbals. The EQ section worked really well here and the mid-band seems to be set at around just the right area for pulling some of the boxy low-midrange area out of a source. Feeling suitably encouraged, I also tried the plug-in on guitars, vocals and keyboards and in one particular mix, several instances of the 67A were hugely important in coaxing a more 'authentic' sound from a collection of instruments that had been recorded in different locations. Although I could hear useful tonal differences when auditioning the options, I rarely found myself straying from the default Tube or Bias settings.

When it comes to saturation, I have a very well-stocked plug-in folder. But a few weeks on from getting the Omnitec-67A I'm still instinctively reaching for it when I crave a bit extra from a source. It certainly does do more extreme personality changes if you want that, but in the middle of a busy mix I've found myself using it for subtler enhancement, where it allows me that bit more control. In that scenario, Having the onboard 'vintage' EQ makes it perfect for adding a bit of sheen to a lead vocal/ acoustic guitar or adding a bit more low end or a kick drum.

Summing Up

As a wider point for discussion, it still impresses me that plug-in designers can recreate authentic signal flows but, ultimately, the user cares about sound and functionality. As a saturation device, this might be better with automatic gain compensation and high- and low-pass side-chain options, but even without that I liked the Omnitech 67A a great deal. It adds some very useful extra features to a great-sounding vintage valve preamp emulation. It's my first encounter with Black Rooster's plug-ins and I'm impressed — and I'm looking forward to trying more.

summary

A great-sounding vintage valve preamp emulation that makes for a lovely saturation effect, and offers some useful EQ options.

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Acustica Audio Silver

Reverb Plug-in

The masters of convolution have finally produced a reverb, but it's far from being a straightforward convolution reverb.

SAM INGLIS

When we hear the word 'convolution', we half expect it to be followed by the word 'reverb'. But the developers who have done the most to push the boundaries of convolution have, until now, used it for practically everything except reverb. There have been reverb IR libraries for Acustica's Nebula, and the Ebony channel strip plug-in includes a reverb, but Silver is Acustica's first dedicated reverb plug-in.

The Silver plug-in itself is free, and comes with four presets to show off what it can do; Acustica aim to make their money by selling additional Volumes of content. However, these are not conventional impulse response libraries, and Silver is not a conventional IR player.

In developing Silver, Acustica identified two long-standing problems that, they feel, afflict convolution reverb in general. One is that basic convolution can only capture a 'static' impression of a space or other reverb generator; it can't recreate the subtle variations, modulations and non-linearities that contribute to the sound. The other is that acoustic and electro-mechanical sources of reverb are, by their very nature, imperfect. Even halls that we perceive as sounding good exhibit resonances and other behaviours that can become problematic when IRs are captured and applied to dry sounds.

To combat both of these tendencies, Acustica have developed a proprietary technology that combines machine learning and conventional DSP techniques.

This applies what they call 'micro-equalisation' to correct resonances and imperfections as they occur, and 'crossfeed' to add motion to the sound. Although details are scant, these seem to be paired with a proprietary measurement technique that captures the properties of a source space in a rather different way from conventional IR recording.

A Is For Reverb

At launch, only one Volume is available, containing over 120 samples from 77 different physical spaces. About half of these represent recording studios, churches, concert halls and other environments relevant to music production; the rest are drawn from domestic and other interiors of the sort that might be used in post-production. Volume A takes up some 10GB of drive space and, like the Silver plug-in itself, is downloaded and authorised painlessly through Acustica's Aquarius utility.

The Silver plug-in itself, meanwhile, comes in three variants. As far as I can tell, the only difference between the full-fat Silver and Silvers Lite and Mini is that the latter have progressively fewer EQ options. I'm not sure why this quite justifies creating separate plug-ins, but of course if the EQ uses Acustica's usual convolution-based technology, it might represent something of a CPU burden. The basic interface presents a manageably small number of parameters, mostly controlled by knobs; activating Guru mode makes visible some less-used controls, but it never gets too crowded or intimidating.

Like algorithmic reverbs, Silver allows for the separate manipulation of early and late reflections. So as well as global control over the wet and dry signal level, there are also large dials for the relative levels of these contributions. Sensibly, all of these retain their settings when a new space is loaded. It's also possible to vary the duration of the late reflections between 50 percent and 150 percent of their 'true' decay time, but the most intriguing controls are the two small »

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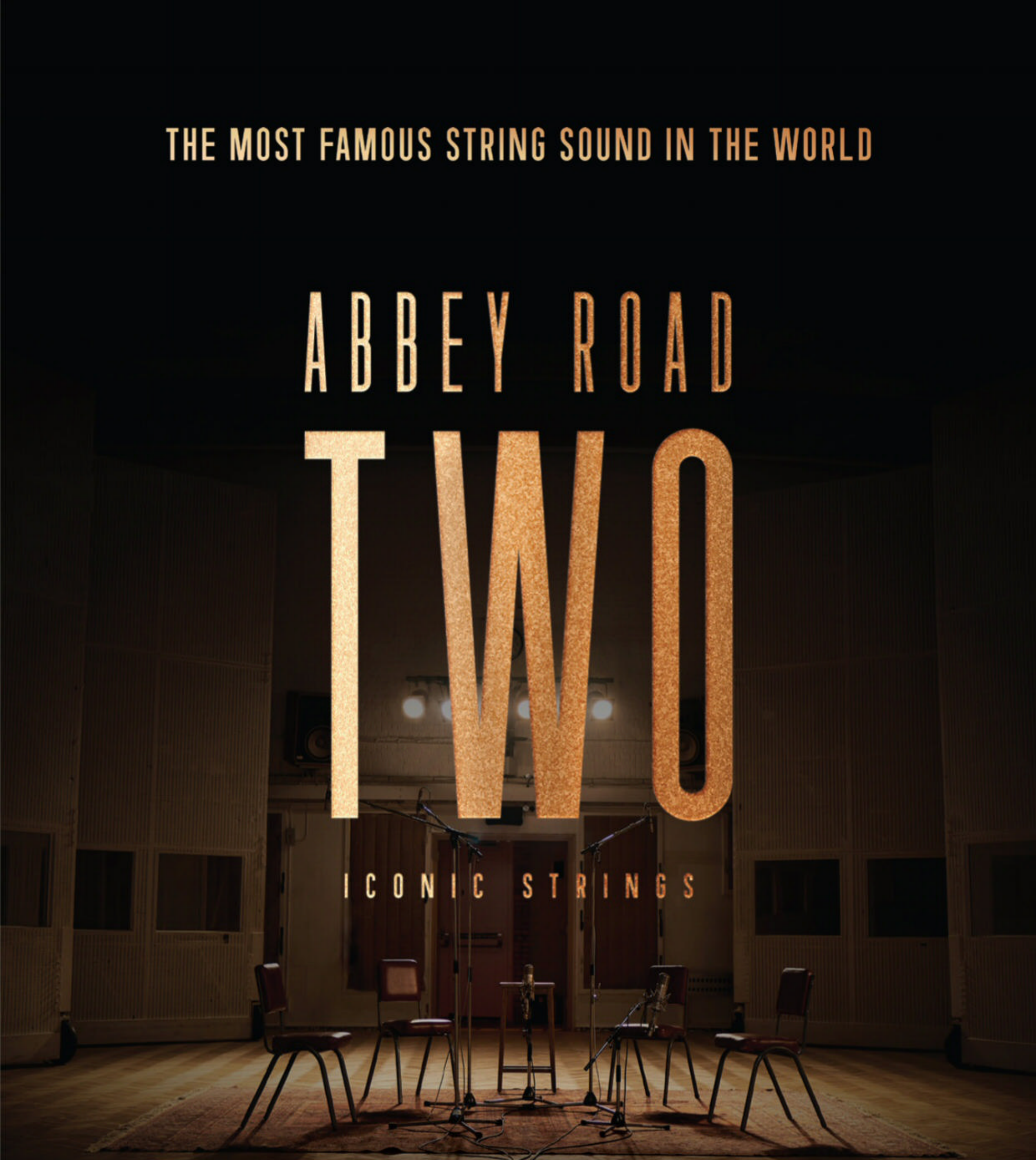

A sample library, created in partnership with Abbey Road Studios.

Abbey Road Studios

SPITFIRE AUDIO

» sliders labelled Perfection. Again relating to the early and late reflections, these set the extent to which they should be 'linearised' or micro-equalised by Acustica's whizzy processing.

The labelling of the three other main controls is a little confusing. Depth is what I would usually consider a Width control, adjusting the level of the Sides component to push the reverb outwards. Width and Pan 3D, meanwhile, are part of the Guru parameter set, and govern crossfeed delay and gain settings within the late reflections part of the signal. All of them affect the 'stereo-ness' of the signal in noticeable ways, Pan 3D being the most obvious, but none seems to act as a straightforward mono-at-one-end-super-wide-at-the-other parameter.

Beyond that, there's relatively little user control other than separate pre-delay settings for early and late reflections, a slightly redundant Pan control and the EQ, which is extremely simple even in the full version of Silver. Other users have reported CPU load to be quite intensive, but this wasn't obvious on my machine, so I rarely bothered loading the 'lite' versions. Artefacts and glitches are noticeable if you adjust parameters during playback, though.

Silver Service

There's often a lot of hype about new plug-ins, especially where some sort of technological breakthrough is claimed. This isn't always justified by the results, but having tried Silver, I am convinced that it does have something different about it. Compared with other artificial reverbs, both algorithmic and convolution-based, it simply sounds less like reverb that's been added after the fact, and more like something that was recorded with the source. For rock and pop mixing, I quickly gravitated towards using the spaces captured in studio live rooms — on drums, acoustic guitars, electric guitars, keyboards and in fact almost anything. Unless you really crank these up, they often don't make the source sound noticeably reverberant at all; their effect in small doses is almost subliminal or subconscious, but real nonetheless. It sometimes feels as though you've simply moved the instrument into a different room.

The Silver Mini plug-in is functionally identical to the full version apart from its EQ options, but takes up less screen space.

If you like to lean heavily on ambiences and subtle room-type reverbs, then, Silver has the potential to make a big difference to the quality and naturalness of your mixes. Move up in scale to the churches and theatres, and what you really notice is that it's easy to get a balanced sound across the frequency spectrum. I never found myself in that situation where you're having to use narrow-band EQ to stop something booming or ringing out in the reverb. In fact, I rarely found I wanted or needed to use EQ at all with Silver. However, one thing to be aware of is that if you reduce the default pre-delay time to zero, the early reflections arrive instantly rather than after a few milliseconds, as they would in real life. In some circumstances, this can cause unwanted comb filtering and phasiness.

The action of the Perfection sliders is interesting. The manual states that the leftmost or minimum setting delivers the 'unimproved' or raw sound of each space, but in practice I often found this to resemble a lo-res MP3. I suspect that Acustica's processing creates some sort of model or synthesis of the space rather than applying IRs directly, and that in its raw form this tends to sound unsophisticated, or perhaps even emphasises unwanted resonances and so on rather than suppressing them. At any rate, there's a good reason why both sliders default to the halfway position! Further increasing Perfection from there reduces harshness and honking, but it also does more than that. Applied to the early reflections, it seems to further ameliorate any comb filtering that can result when dry and wet signals are recombined, and to stop transients sounding splashy. On the tail, meanwhile, it smooths things out, counteracting any tendency towards 'tearing' or 'barking' (for want of better words) and sometimes slightly reducing the apparent decay time by curtailing ringing. Higher Perfection settings also seem to preserve the high end more faithfully. You could perhaps achieve a little of the same effect by placing a plug-in like Oeksound's Soothe or Soundtheory's Gullfoss in series with a reverb, but it certainly wouldn't be the same.

There's clearly huge potential here, then, but it should be noted that Volume A doesn't cover all bases. It's fantastic for natural-sounding rooms, and there are some lovely churches; but studio staples such as plates, springs and chambers are absent, and most of the hall-type spaces seem oriented more towards post-production than music. I think you'd struggle to put together a convincing pop vocal treatment using the Volume A spaces alone — but that's not really a criticism because, as the name suggests, more Volumes will be forthcoming. In fact, just as we went to press, Acustica announced Volume B, a library of plate reverbs for Silver. If they can improve on real plates in the same way that Volume A does on real spaces, then Silver could be a first-choice reverb in every role.

summary

Silver is capable of integrating source and reverb with a natural quality that even other convolution plug-ins struggle to match.

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Synth Gems 1

Book Review

The latest lavish offering from Bjooks collects some of the world's rarest and most desirable synths between its covers.

ROB PURICELLI

One of the best things about researching synthesizer history these days is the Internet. Equally, one of the worst things about researching synthesizer history these days is the Internet. The same can be said for a great many things, because, just like a signal booster, while the Internet increases the good bits of a signal, it also amplifies the bad. There is a desperate need for a filter and, when it comes to knowledge, a respected author with an almost clinical attention to detail and accuracy is required. Luckily for us, Dr Mike Metlay is the Moog Ladder to our Noise Generator. After months, if not years, of painstaking research and collaboration, he has delivered Synth Gems 1, the next in an ever-growing line of superb synth books from the highly respected Bjooks stable.

What we have here is a collection of synthesizers that run the gamut of rarity and uniqueness, many of which you will know of, and many of which you will not. Around 50 instruments come under Mike's scrutiny, each chapter detailing how the instrument worked, how it came into existence and the impact it had. Accompanying Mike's words are some of the most exquisite photographs you are ever likely to see of these machines, expertly capturing their visual impact in superb detail. Examples range from the obvious (Moog Minimoog, Sequential Prophet-5, Fairlight CMI, Roland Jupiter 8) to the rare (Yamaha GX-1, ARP Quadra, OSC OSCar) and mostly unheard of (Gleeman Pentaphonic Clear, EML SynKey and Elgam Carousel). And just like any collective list, there will be examples you agree with and others that you don't, but Mike sets out to justify his choice with every example and does so with aplomb.

After a foreword by the great Vince Clarke, the book starts with an introduction to 'the synthesizer' which far from being a patronising section to those of us with decades of experience, is a superb introduction to the uninitiated and newcomers to electronic music making. And I'd wager that even the most long in the tooth synth heads could learn a thing or two, or at least learn how to better articulate such concepts. This is complemented by a useful glossary at the back of the book.

We are then taken on a chronological tour of the gems, as Mike sees them, and I was sometimes pleasantly surprised not just by the choices, but some of the specific models chosen. For example, the book opens with the Moog Minimoog. I don't think anyone can argue with that choice. However, it is not the Model D that hogs the limelight here — the preceding models A, B and C are equally represented alongside one another, offering the reader a rare glimpse at the lineage and progression that Moog made with this archetypal instrument.

It is also refreshing to see that alongside such behemoths as the CS-80, Waldorf Wave and Andromeda A6, pivotal gateways into synthesis also get their moment in the spotlight, such as the Casio VL-Tone, EDP Wasp and Ensoniq's ESQ-1 and SQ-80.

If all of that wasn't enough, the book also features detailed appendices that contain details of the resources that helped provide the instruments covered in the book, as well as links to preservation establishments, online groups and further reading/viewing, including this very magazine! And knowing that many of these instruments are, for all sorts of reasons, well out of reach for many of us mere mortals, there is a list of software recreations which refreshingly flies in the face of the inherent gear snobbery that often permeates the world of vintage synthesizers.

Swamped, as we are, with so much unfiltered information about so many things, books like Synth Gems 1 are more vital than ever in our noble pursuit of the truth about the wondrous technological achievements found in the world of synthesizers. It is also a source of nerdy dreams and wallet-emptying gear lust!

summary

A unique publication that serves as an historically accurate document, both written and visual, and will, I have no doubt, find itself on the shopping lists of the synth nerds amongst us.

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Eventide TriceraChorus

Chorus Stompbox

Another of Eventide's excellent plug-ins makes the transition to the popular pedal format.

PAUL WHITE

Although all of the effects in this pedal are created digitally, the sonic inspiration for the TriceraChorus lies in bucket-brigade analogue chorus pedals from the 1970s and '80s. The chorus effects are combined here with Eventide's MicroPitch detuning and a parameter called Swirl, which adds what Eventide describe as psychedelic flanging, phasing, and Univibe-style sounds. Furthermore, TriceraChorus can generate three different types of chorus voice with separate modulation depths for Left, Center and Right, to generate a lush stereo soundscape. There are three basic chorus types designated as Chorus, Vibrato, Chorale and up to 200ms of delay may be added to each type.

Overview

As with Eventide's UltraTap Delay and MicroPitch Delay, which share the same form factor, the TriceraChorus has the ability to store presets. In this case up to five can be stored in standalone mode, but the free Mac/Windows Eventide Device Manager (EDM) software gives access to many more and provides a means to create new presets too. Five LEDs show which preset is active and up to 127 presets can be accessed via a MIDI connection.

The pedal itself is made from folded steel and comes with five card overlays, each showing the control settings for a different factory preset. There are six knobs, each of which has a dual function, and two footswitches with status LEDs. Active is used to engage or bypass (which can be set as Buffered, Relay, DSP+FX or Kill Dry) the effect, and one activates or bypasses the Swirl option, with a button switching between the two Swirl knob functions. The Swirl footswitch can also be used to select presets, while the Active footswitch can be set for latching or non-latching operation.

The main knob functions are Mix (which also selects the modulation type), Rate, Detune, Left, Centre and Right, with secondary functions of Mix Envelope, Rate Envelope, Pitch, Delay, Filter and Output Level. The envelope settings allow the input signal envelope to control aspects of the modulation, for example, allowing the modulation depth to increase as the input sound decays.

On the back are an input for the included 9V PSU, a mini-USB port for linking to the software or MIDI control over USB, and quarter-inch jacks to accommodate mono or stereo inputs and outputs. The input is on a TRS stereo jack, and the two outputs appear on separate mono jacks, with switching for instrument or line-level operation and separate mono/stereo switching for the inputs and outputs. A further jack takes an optional expression pedal or switch, for tapping in the mod rate, and using a three-button switch here you can cycle through presets. Any combination of parameters can be mapped to the expression pedal to allow smooth real-time morphing between settings.

Sound Impression?

Of course, what really matters is the way this thing sounds, and it really is very impressive in this respect. If you've heard Eventide's plug-in equivalent of this pedal, you'll already know what to expect: it covers all of the conventional chorus-like effects but it can manage a credible ADT or doubling effect too, with the detune feature helping reinforce the illusion. Kicking in Swirl really does suggest a hint of Univibe, and the stereo options allow for rich, textural, almost '3D' chorus effects. Importantly, this pedal isn't just for guitarists: the TriceraChorus sounds just as impressive on keyboards. It really doesn't disappoint, either in the range of available modulation effects or in its overall sound quality.

summary

A great-sounding, high-quality digital chorus pedal, suitable for line/instrument and mono/stereo sources.

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Toontrack Fields Of Rock SDX

Expansion For Superior Drummer 3

For their latest SD3 expansion Toontrack made a pilgrimage to the hallowed ground of Rockfield Studios.

MARK GORDON

The latest addition to Toontrack's extensive Superior Drummer 3 drum and percussion library (currently comprising 18 titles) is Fields Of Rock SDX. Recorded at the legendary Rockfield Studios in Wales, Fields of Rock features seven kits and a comprehensive set of percussion instruments including rototoms, octobans, congas, shakers and more, all recorded by multiple award-winning engineer/producer Tom Dalgety (Royal Blood, Pixies, Killing Joke and Rammstein) and played by session drummer Mat Hector.

Field Of Dreams

In 1965 Rockfield became the world's first residential studio, and it has witnessed the birth of some of the most iconic recordings of the last 50 years, by bands ranging from Queen, Motörhead and Black Sabbath to Oasis and Coldplay. Housed within a working livestock farm, the studio comprises two rooms: the Coachhouse and the Quadrangle. Both of them were used to record the three drum libraries and additional percussion library that make up the Fields Of Rock SDX.

The Coachhouse is based around a Neve 8128 inline desk, with outboard including Neve 1060 microphone amps. Its heavily draped recording area provides the location for the 'Dry Room' library. Kits were also set up in front of the exposed stone wall for the more lively 'Stone Room' library. The 'Live Room' library was recorded in the Quadrangle (in which Queen's Bohemian Rhapsody was also recorded), which features an MCI 500-series inline desk and has hardwood flooring and more reflective surfaces, providing a contrasting ambience to the Coachhouse.

No Silence In The Library

The Dry Room library features three kits: two 1970s Ludwigs and a Hayman Vibrasonic from the 1960s. The kits used in the SDX libraries often have a story to tell, and in this instance drummer Mat Hector used both Ludwig kits during his time with Iggy Pop. The Ludwig 400 Supraphonic snare that features in this library is an all-time classic that has probably provided the backbeat for more classic songs than any other drum.

The Hayman kit is heavily damped and can be routed through a series of outboard guitar effects that enables it to live up to its nickname, the 'Evil Disco Kit'. All three kits offer a deep and warm tone and a definite '70s vibe — but with plenty of attack — which is complemented by a range of dark Istanbul cymbals.

The Live Room library features a single kit from the British Drum Company, dubbed the Stormtrooper on account of its bright white finish. The ambient room mics allow for a big, open drum sound which, although the kit was built in 2020, lends itself to the late-'80s/early-'90s arena rock sound. It's partnered with large, bright cymbals and a Van Kleef cast-bronze snare drum that suit the kit perfectly.

The Stone Room library goes from the sublime to the ridiculous with its selection of kits. At the 'normal' end of the spectrum is a Ludwig classic oak kit, which has a modern sound that will fit into almost any style of music. Like the Stormtrooper, this kit was also built in 2020 but offers a warm, vintage sound that is in keeping with the library as a whole.

Moving a little further leftfield is the 1980s Sonor Sonorlite, a beast of a kit featuring seven deep birch 'power' toms. Nicknamed the 'Pure Goth', this kit channels 1980s darker rock acts, such as the Cure and Siouxsie & the Banshees, and allows for huge drum fills and inventive patterns to be played across all those toms.

Last up is the 'Futuristic', a 1980s Staccato kit with fibreglass shells. If you've ever seen one of these kits, you won't have forgotten it! The single-head concert toms flare out in dramatic fashion and »

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» the bass drums look like a pair of shorts for an elephant. Fibreglass shells and single-head toms give a very punchy and focused sound, reminiscent of classic ’80s Phil Collins. Gate the ambience mics and you could almost create ‘that’ drum fill.

Finally, we come to the percussion library. In addition to the selection of congas, bongos, tambourines and shakers that you might expect, Toontrack have included a set of Rototoms and Octobans, which also featured in the exceptional Rooms Of Hansa SDX. But it doesn’t end there. ‘The Junkyard’ includes a whole range of intriguing Foley pieces that have been hit, dropped and shaken in various ways, to create a very interesting set of sounds. You’d be amazed by the sort of rhythms and grooves you can create from some window railings, a cutlery drawer and a step ladder (single-rung hits, glissando up and down the steps, and kicked across the floor... obviously!).

We Like How They Mic

As with all Superior Drummer libraries, microphone placement plays a big part in creating the overall drum sound. The Fields Of Rock SDX is no exception, with the close-miked drums augmented by a range of mono and stereo ambient mics to add the room acoustics. To capture the unique sound of the Coachhouse, the Stone Room library includes a Shure SM57 placed in the corridor between the two studios for an additional ambience that, when mixed with the whole kit, is particularly apparent on the snare. The unique Rockfield reverb chamber is also featured in this library, as well as a mono microphone routed through a selection of outboard guitar effects pedals to add a ‘trash’ quality.

The Junkyard Percussion section includes a number of things not previously recognised as musical instruments. The step ladder, for example...

Groove Digger

In addition to the ‘raw’, unaffected kits, Toontrack provide a number of presets created by Tom Dalgety and Toontrack themselves. These feature unique combinations of drums and processing that open up a wealth of creative techniques and options to explore.

A selection of MIDI grooves, played by Mat Hector, is also included. Divided into sections that relate directly to each library, the grooves offer a broad insight into how Tom Dalgety sees the kits being used. For example, the grooves suggested for the Stone Room’s Pure Goth kit have a definite air of Budgie from the Banshees about them.

There are verse, chorus, bridge and fill grooves that cover a multitude of styles and feels. Of course, you can use these or any Superior Drummer 3 grooves with any kit from any library, to create the perfect drum track, and further edit any element of it using the grid editor within SD3.

‘Futuristic’ — a Staccato kit with fibreglass shells.

Bang On

Field Of Rock’s three contrasting spaces and range of kits give a wide and varied sound palette, which, although well suited to ’70s and ’80s rock, also fit very well into more modern genres. The percussion is a fantastic addition to the drum libraries and is extremely usable.

If you need help to decide whether this library is for you, the Toontrack website provides the most comprehensive detail you could wish for, with every aspect of the equipment, rooms and recording process covered in incredible detail. In fact, it makes a very entertaining and informative read in its own right!

This is not a library of extremes — there are no huge spaces or complex surround mic techniques to contend with. A little like the studio in which it was created, this is a highly practical and functional library with a lot of character and a definite whiff of history about it.

summary

The Fields of Rock SDX offers a fantastic range of kits that capture the essence of 70s and 80s rock with a contemporary edge.

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Avantone Gauss 7

Active Nearfield Monitors

Avantone's newest monitors revive a classic speaker brand and channel the '70s West Coast studio sound.

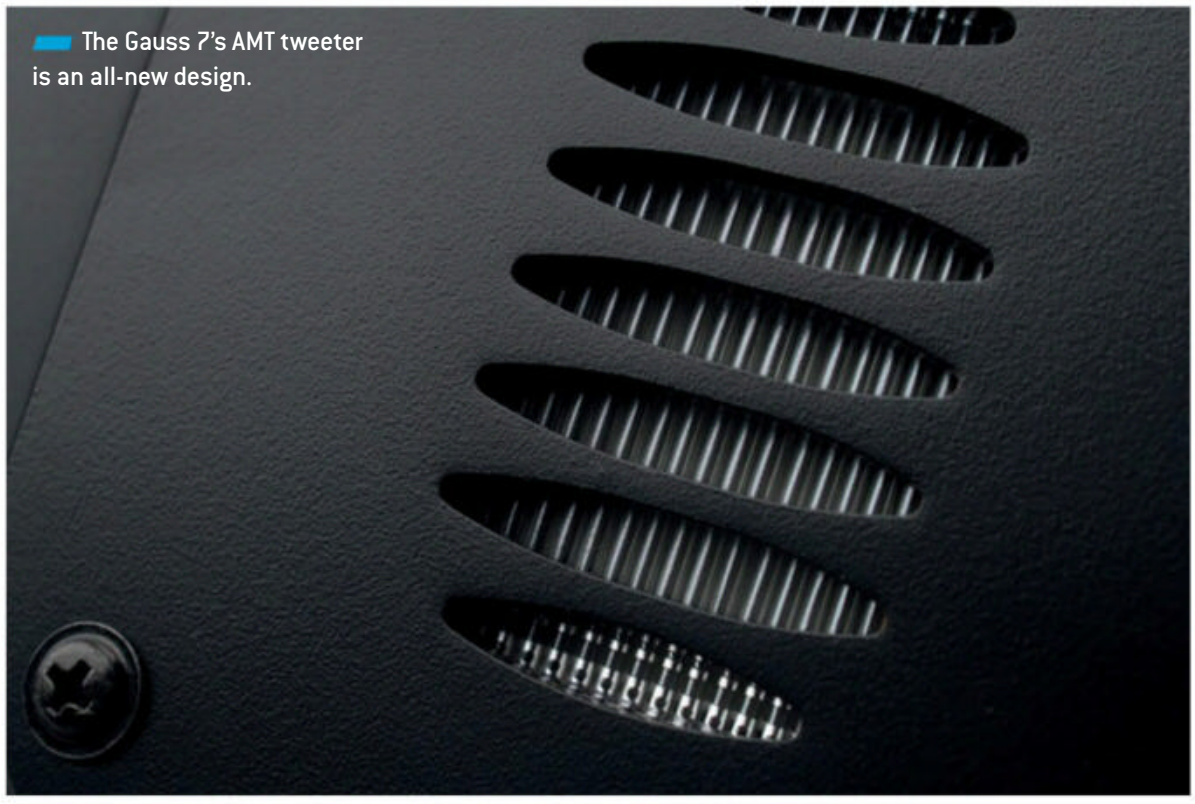
PHIL WARD

Avantone Pro appear to be creating a business out of mining the monitor archive. First it was their MixCube, an homage to the Auratone single-driver compact monitor; then their CLA-10 clone of the Yamaha NS-10; and now the venerable old American speaker brand, Gauss, has found itself brought back to life under the Avantone umbrella. However, Avantone's new Gauss 7 monitor bears no relation to products by the original Gauss company — Avantone have simply purchased the rights to the name and are planning a range of monitors that will use it.

Before I describe the Avantone Gauss 7 though, I'll cover a little of the history of the original Gauss company. I write "a little" because it turns out that the Internet has a bit of a blind spot where Gauss are concerned, and there's not all that much history to be found. It's not even clear to me if the name, Gauss, was inspired by Carl Friedrich Gauss, the German physicist who gave his name to a unit of magnetic flux, or if that's just a random coincidence. The basic story, however, is that Gauss were founded in the 1960s by some ex-JBL driver engineers. They launched a range of 10- to 18-inch-diameter drivers and horn-loaded compression tweeters, aimed at PA, guitar backline and US West Coast-style main studio monitor applications. Gauss also subsequently launched some complete, JBL-inspired main monitors. They gained a reputation for engineering advances over their immediate competitors by introducing such innovations as double spider suspensions, direct-wound aluminium voice coils and formers, and braided voice-coil lead-out wires. These are all relatively common features today, but

back in the 1960s they were new. Gauss were sold in the 1970s to the parent company of Electro-Voice and Altec, and then all three brands were sold on to the Telex manufacturing group (now part of Bosch), at which point the Altec and Gauss products were integrated into the Electro-Voice range and the Gauss name in particular disappeared.

Despite the fact that there's no direct link between Gauss now and Gauss back in its heyday, the philosophy of Avantone's new Gauss 7 speaks of the values that are often identified with 1970s studio monitors: high electro-acoustic efficiency and snappy dynamics prioritised over a ruler-flat frequency response and low coloration. Having said that though, there wasn't much alternative to high-efficiency and somewhat unrefined tonal qualities when amplifier power was more limited than it is today and driver diaphragm and enclosure materials were relatively undeveloped. West Coast monitors of the 1970s were, in these terms, pretty much as far away as it is possible to get from, for example, the BBC monitor sound of around the same time.


The Gauss 7's AMT tweeter is an all-new design.

Knocking On Seven's Door

So, to the Gauss 7 itself. It's a relatively plain-looking yet neatly detailed two-way nearfield monitor, of dimensions that ought to cause no problems in the vast majority of studio rooms — even small ones. The cabinet proportions quite usefully eschew the current trend for extended depth, which can only help with installation in cramped spaces. The attractively angular profiled and satin-black painted MDF front panel carries a nominally 180mm (7-inch) bass/mid driver and a 30 x 55 mm Air Motion Transformer-style ribbon tweeter. Behind the front panel, the cabinet sports a 'black ash' veneer finish, and towards the bottom left of the front panel is an LED that illuminates in red and blue to indicate standby and normal operation respectively.

The Gauss 7 bass/mid driver is noticeable of course for its white paper diaphragm, which brings to mind not only the NS-10 and the later Yamaha HS Series of monitors but also Avantone's own NS-10 clone, the CLA-10. However, while the Gauss 7's white diaphragm shares its paper pulp formulation with the CLA-10 driver, Avantone Pro say it's a completely different driver design, with a stiffer diaphragm, larger magnet system and a butyl rubber surround engineered to enable significantly greater diaphragm movement.

The Gauss 7's AMT tweeter is a fresh design, Avantone say, rather than an off-the-shelf item. AMT tweeters differ from pure ribbon devices by employing electrical conductors 'printed' on a concertina-folded substrate rather than using the (usually flat) aluminium ribbon itself as the conductor. In both cases the conductor is held between the poles of a magnet so it moves in response to the electrical signal. The primary advantages of the AMT devices are that, thanks to the length of the conductor, it doesn't require the input transformer that a pure ribbon needs for impedance matching, and thanks to the large, effective radiating area that results from the concertina folding of the diaphragm, it's much more efficient. AMT disadvantages are that, in comparison to a pure ribbon, they have significantly higher moving mass (although it's usually still substantially less than a dome tweeter), and because of their physical diaphragm size they tend to display somewhat restricted high-frequency dispersion. If you've had one eye on the nearfield monitor market in recent times you will probably have noticed the increasing popularity of the AMT ribbon, and there's little doubt that they can be made to work extremely well.

Around the back of the Gauss 7 there's a connection, control and heatsink panel. As far as connections are concerned, the Gauss 7 offers analogue balanced inputs with options of TRS jack and XLR sockets. There's no digital input option. Controls comprise a ±6dB variable gain knob, and switches that engage flat or ±2dB high-frequency shelf EQ options; and 0dB, -2dB or -4dB room boundary compensation LF EQ. The -2dB and -4dB options are suggested for wall and corner installations, respectively. Before I leave matters of Gauss 7 control, if you've read many of my reviews you'll know that I can get grumpy over variable gain knobs, because they introduce doubt over monitor level pair matching. Avantone Pro do actually acknowledge this in the Gauss 7 Quick Start Guide, and suggest that a sound pressure level meter or sound level metering app is used to confirm pair level matching at the listening position. I still think knob detents would be preferable though, firstly because that would mean that no messing about with SPL meters is required, and secondly, setting the level using a full-band measure of SPL will potentially be skewed by listening position frequency response »

Avantone Gauss 7

£775

PROS

- Well-judged bass performance.
- Very capable AMT tweeter.
- Good at high volumes.

CONS

- Midrange coloration.

SUMMARY

The Gauss 7 inexpensively channels a vintage monitor vibe with an engaging combination of characteristics — some positive, some... vintage.

» anomalies that result from differences between left and right monitor locations. For example, if one monitor is significantly closer to a corner than the other, that could result in significantly higher SPL at low and low-mid frequencies. If the resulting skewed SPL is then measured and compensated by full-band gain reduction on one monitor, it will then be quieter than its opposite speaker in the upper-mid and high-frequency band. One of the more subtle benefits of room/monitor optimisation apps such as Sonarworks, ARC or Dirac Live is that they can fix pair level matching issues such as these, but I still think knob detents or even switches would be better.

Along with the knobs, switches and sockets, a reflex port is also to be found on the Gauss 7 rear panel. The port is of a healthily large diameter (the larger diameter of a port, the more air volume can flow without turbulence) and fairly generously flared. Due to the relatively short cabinet depth of the Gauss 7, its reflex port tube takes a sharp 90-degree bend downwards internally so that the length required to achieve the appropriate tuning frequency can be made to fit. Port tubes with bends are not unusual so the Gauss 7 is by no means alone in having one, however sharp bends do result in the earlier onset of turbulence. Having said that, bends can also suppress the likelihood of undesirable organ pipe resonance. I'll investigate the Gauss 7 port with a little FuzzMeasure analysis...

Call The Fuzz

And speaking of FuzzMeasure analysis, Diagram 1 illustrates the Gauss 7's axial and 20 degrees vertically off-axis frequency response, the latter measured both 20 degrees upwards and downwards. The main thing to notice is that the response through the crossover region between 2kHz and 5kHz is somewhat untidy. And where I'd normally expect one or other of the vertical off-axis curves to display the results of driver overlap interference, in the case of the Gauss 7, both show discontinuities. Having said that, however, when measuring the Gauss 7 I found it was particularly sensitive around the crossover region to microphone position, which suggests that the bass/mid driver response is somewhat variable at the top end of the range, making managing its integration with the tweeter somewhat hit and miss. Such a characteristic is not at all unusual with paper-diaphragm drivers, especially those engineered for high sensitivity and high volume level potential, as I suspect the Gauss 7 driver is. The second characteristic revealed by Diagram 1 is that, as expected, the AMT tweeter shows relatively restricted vertical off-axis dispersion.

Diagrams 2 and 3 reveal some characteristics of the Gauss 7's low-frequency performance. The first of the two diagrams illustrates a close-mic frequency response of the bass/mid driver. And I mean really close; the mic was less than a centimetre from the diaphragm. Make such a measurement with a closed-box monitor and you'll see a pretty accurate measure of its low-frequency roll-off, and although that doesn't work with a ported speaker, the technique still offers some useful information. The monitor's port-tuning frequency in particular is revealed by a sharp suck-out. In the case of the Gauss 7, it's at 50Hz, give or take. 50Hz is pretty typical for a monitor of the Gauss 7's dimensions, but what is less usual is that the suck-out is relatively wide and not particularly deep, which suggests that the port resonance is well damped.

Diagram 3 illustrates the close-mic response curve of the reflex port itself. There's two things to note. Firstly, the port output extends significantly either side of the 50Hz tuning frequency, which confirms that the port Q is relatively low »

The rear panel hosts the analogue audio inputs, an input gain knob, HF trim switches and boundary compensation EQ controls.

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» and that the Gauss 7 falls more into a 'damped reflex' category than a 'squeeze as much bandwidth extension as possible out of the port' category. To my mind this augurs well for the Gauss 7's subjective low-frequency performance. The second thing to note is not quite so positive and it's that, despite the sharp bend in the port tube, the Gauss 7 does appears to suffer some organ pipe port resonance, as revealed by the peaks in the curve at 950Hz, around 1.5kHz, just under 2kHz and just under 3kHz. This series suggests the peaks are harmonically related. The most troubling resonance is the one at 950Hz because not only does it peak at only 8dB below the port's low-frequency output, it's located at a frequency where the ear is extremely sensitive. I'd expect it to make a contribution to the Gauss 7's subjective signature.

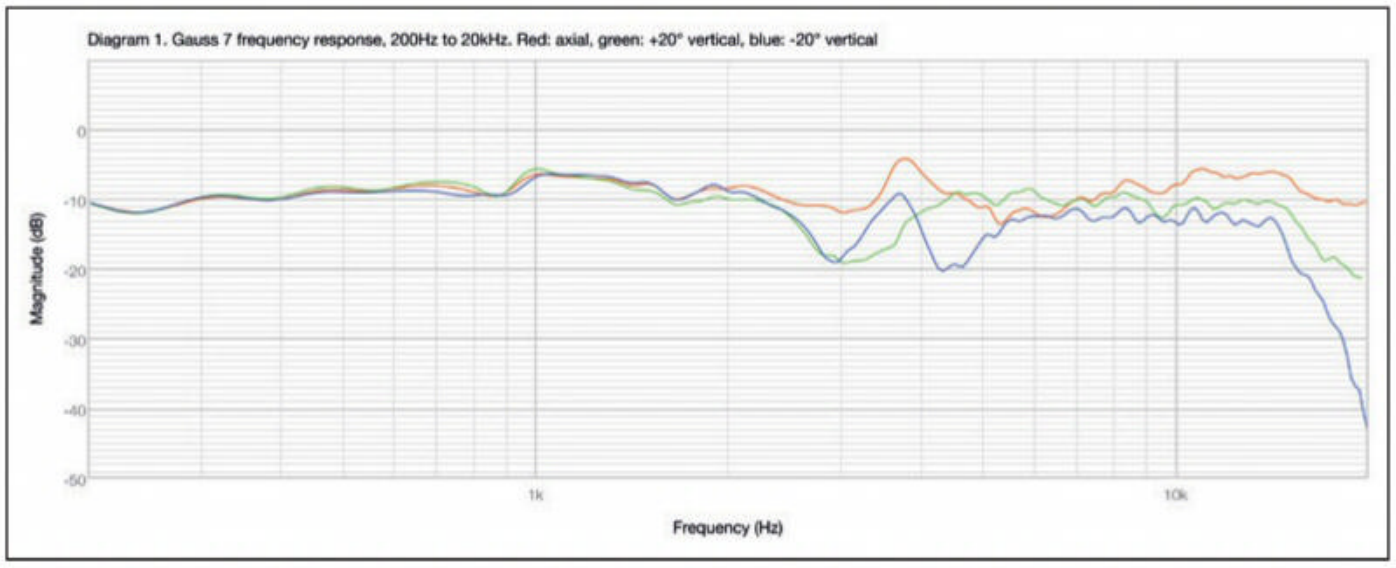

Diagram 1: The Gauss 7's on-axis frequency response (red trace), plus measurements taken 20 degrees above and below axis (green and blue traces, respectively)

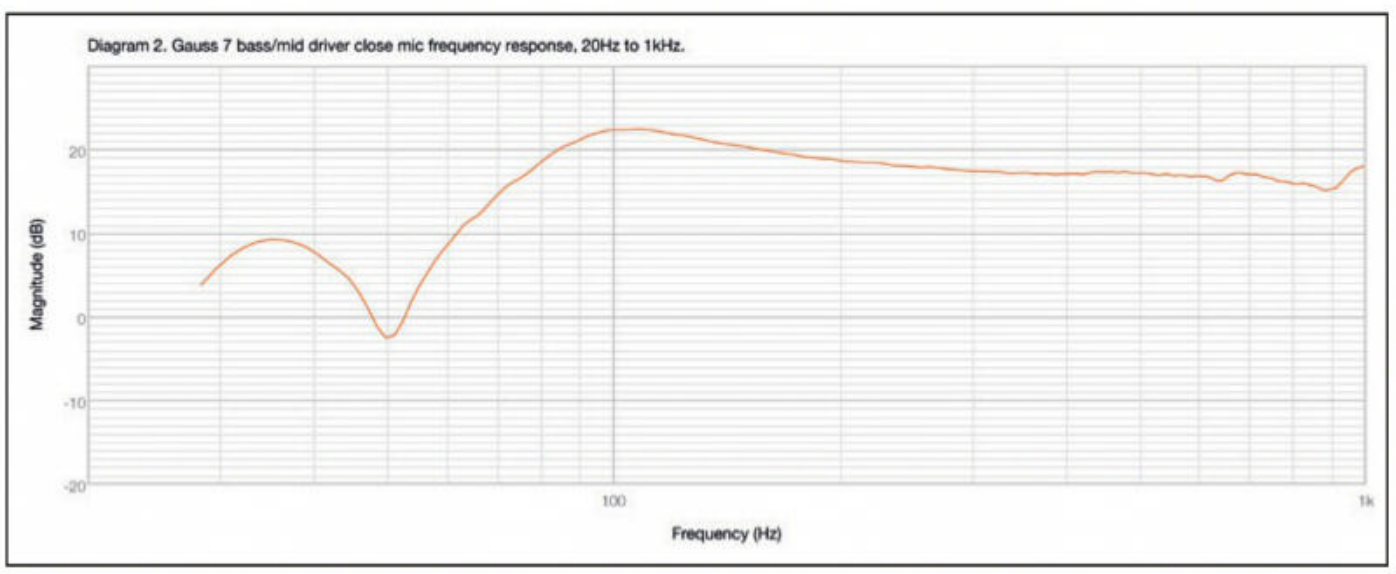

Diagram 2: A close-mic measurement of the bass driver reveals a port-tuning frequency of 50Hz.

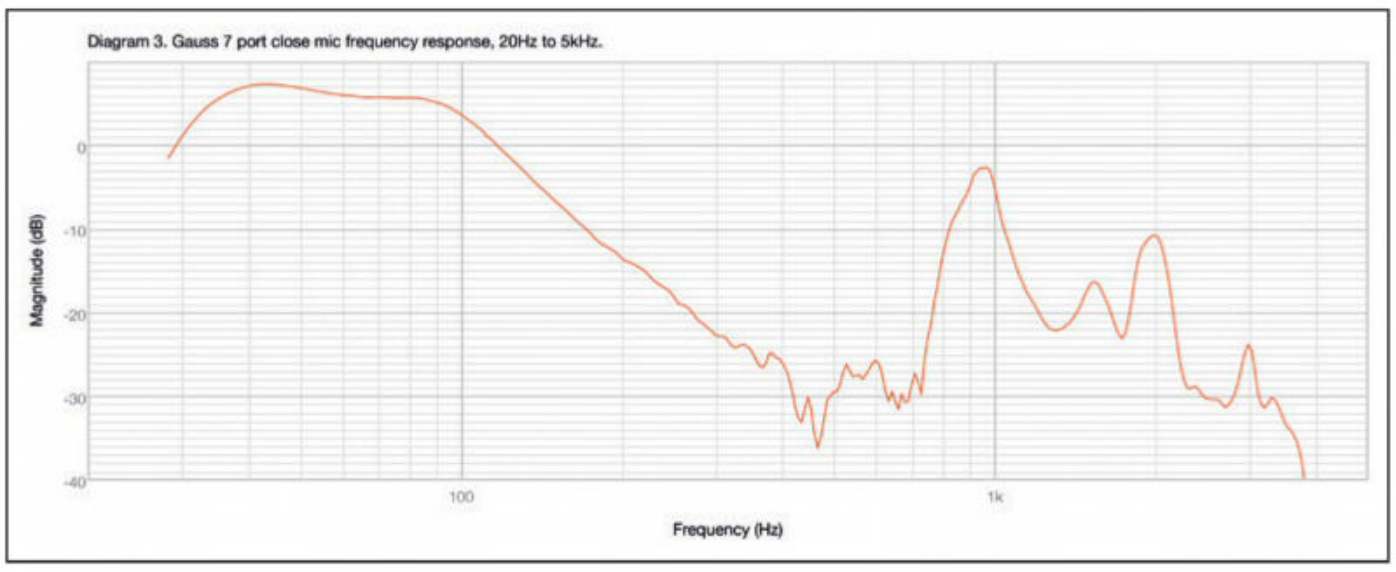

Diagram 3: A close-mic measurement of the Gauss 7's port.

To illustrate the port resonance in a little more detail, I generated a waterfall plot of the close-mic port output, and that's illustrated in Diagram 4. If you've not come across a waterfall plot before, it illustrates how a frequency response decays after the input signal stops. Resonant features are revealed by ridges running from the back to the front. In the Gauss 7 waterfall plot, the major low-frequency port resonance dominates visually, but the 950Hz feature in particular is also unmistakably apparent.

The final FuzzMeasure curve, Diagram 5, illustrates data I've not presented previously in monitor reviews. The frequency response curves displayed are generated by attaching an accelerometer to the side panel of the enclosure to visualise how much, and at what frequencies, it vibrates in response to an input signal. Enclosure panel vibration is important because it can make a significant detrimental contribution to the subjective character of a monitor.

The Diagram 5 curves show the cabinet accelerometer outputs of the Gauss 7 and of a BBC LS3/5A, with both monitors generating the same broadband acoustic output level. The differences aren't insignificant and the Gauss 7 clearly has a less rigid and more resonant cabinet than the BBC LS3/5A — especially in the two octaves from around 400Hz to 1.5kHz where the ear is most sensitive and where so much musically critical information lives. Of course, in some respects, the comparison is unfair because an LS3/5A is not only a far more expensively engineered monitor, it was also specifically designed to reproduce acoustic sources as accurately as possible. Having said that, it's also the case that the LS3/5A was designed not far short of 50 years ago, so it's not as if the technology and techniques needed to minimise cabinet vibration are not well known.

Listening In

Despite the fact that I most often in my reviews write about FuzzMeasure

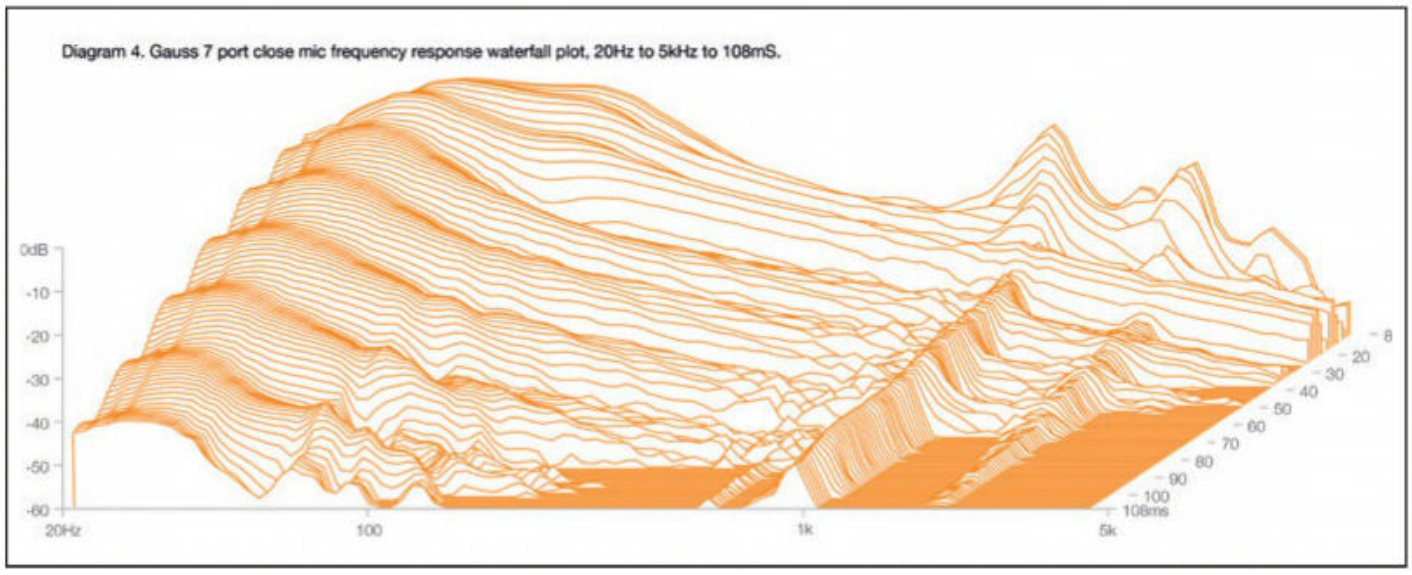

Diagram 4: A waterfall plot measured at the port exit.

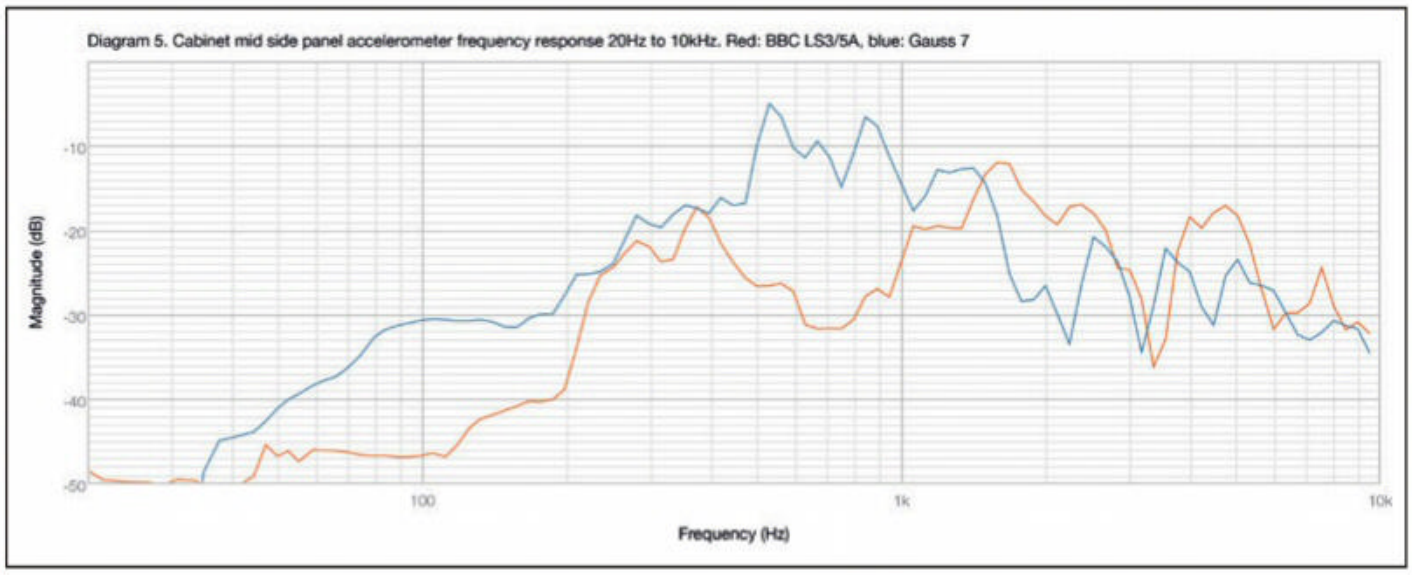

Diagram 5: Measurement of cabinet activity taken with an accelerometer placed in the middle of a side panel. The Gauss 7 is shown in blue, and a BBC LS3/5A in red.

analysis before I describe how a monitor sounds, the actual process happens the other way around. It's important I think that my initial subjective opinion is not influenced by a bunch of objective measurements. And speaking of my subjective opinion, on listening to a variety of CDs and Pro Tools mix sessions, the Gauss 7 revealed itself in general terms to possess a fundamentally usable, if slightly mid-emphasised and bright tonal balance. I'd be reasonably confident of producing mixes that translate in overall balance terms right from the off, however I pretty quickly reached around to engage the rear panel EQ to drop the tweeter by 2dB. It worked better that way in my studio (I also engaged the -2dB LF boundary shelf). The Gauss 7 tweeter, since I'm on the subject, is a success; it is revealing of high-frequency detail without particularly drawing attention to the fact.

Generally the Gauss 7 reveals an impressive level of mix detail, due to some extent to its mid emphasis (NS-10 style), albeit with a slightly flat and unfocussed stereo image quality. And its subjective bass quality is notably impressive for an inexpensive ported monitor. To my ears its designers have hit on a good compromise between bandwidth extension and dynamic character. Hand in hand with its low-frequency performance, the Gauss 7 also impressed with its ability to play at surprisingly high levels without obvious signs of either distortion or compression. In that respect it reflects the original Gauss company's philosophies well.

ALTERNATIVES

The Gauss 7 finds itself in a decidedly crowded part of the monitor market inhabited by many, many alternatives. I'd consider monitors such as the **Kali Audio IN-5** or **IN-8**, the **Focal Alpha 65 Evo**, the **IK Multimedia iLoud MTM** and **Dynaudio Lyd 7** for a start.

However, referring back to my FuzzMeasure analysis, it perhaps won't come as a surprise to read that, along with the positives, I felt that the Gauss 7 is not without some subjective flaws. In particular, it displays to my ears a noticeable nasal and boxy character that colours voices and acoustic instruments. Feed it with less organic mix elements, synths and processed loops, say, and you may well not notice, but listen to naturally recorded voices, woodwind or acoustic guitars, for example, and the Gauss 7 imprints a degree of its own tonal personality on theirs. I suspect there's three phenomena responsible: firstly the FuzzMeasure analysis suggests that the bass/mid driver diaphragm is somewhat uncontrolled towards the top of its band, and that could well result in a noticeable upper-midrange signature. Secondly, the Gauss 7's cabinet panels clearly play along with the music and, again, this is very likely to be audible (when the BBC LS3/5A was developed, even the variety of wood used for internal cabinet corner battens was found to be significant). And thirdly, that port organ pipe resonance is likely to play a part in the overall sound.

I've described some Gauss 7 flaws in that last paragraph, however it's important I think not to get too hung up on them because the context of the Gauss 7 is, firstly, that it's inexpensive, and there's no such thing as a flawless inexpensive monitor, and secondly, that it's intended to some extent to channel a West Coast vintage monitor vibe. And those monitors were flawed too. There's also elements of the Gauss 7 that work really well for the price. Its low-frequency performance is good, it plays loud without trouble and its tweeter works well, so if the vintage vibe appeals, and your mix work doesn't require the last word in midrange tonal accuracy on voices, the Gauss 7 is an appealing option. I probably wouldn't want to mix a BBC radio play or a Mozart string quartet on the Gauss 7, but an Eagles tribute band? Absolutely.

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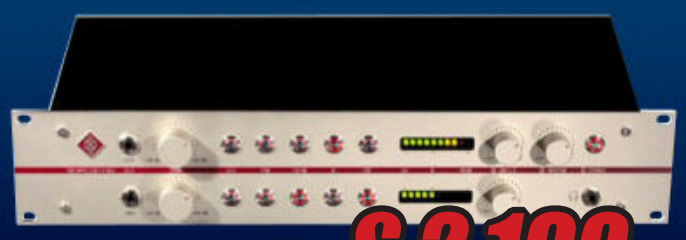

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The Korg Nautilus 61 (Black) is a 61-Note Music Workstation which revisits the idea of what a modern Synthesiser should be able to do. The clearly arranged Control Panel quickly makes it clear that most operations are carried out on the large, colour 7" Touch-Screen. **SYN0007725-000**

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Roland JUPITER-Xm

Roland's Jupiter-Xm is a state-of-the-art keyboard synthesizer that wants to be more than just a reissue of the famous Jupiter-8. With the latest Zen-Core sound generation, sounds of numerous classics like Jupiter-8, Juno-106, JX-8P, SH-101, XV-5080, RD-Piano, TR-808, TR-909, CR-78 are available.
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Roland Fantom 6 (Black)

The Roland Fantom 6 (Black) is a 61-Note Digital Synthesizer Workstation with a Semi-Weighted Keyboard, channel after-touch, Sequencer, Sampler, Audio Interface and extensive Synthesizer functionality. In the expandable sound memory, Roland has created a best-of from 40 years of its own prestigious history to present an Instrument of great power to ambitious musicians. **SYN0007189-000**

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UDO Audio Super 6 (Blue)

The Super 6, a 12-voice polyphonic, digital-analog hybrid synthesizer with binaural voice architecture, was developed by UDO Audio. The robust and high-quality manufactured instrument is equipped with a velocity-sensitive 49-key keyboard with aftertouch by Fatar and stands not only due to the numerous control elements for easy operability and high playing pleasure. **SYN0007131-000**

£3,292

Waldorf Quantum

The Waldorf Quantum is a 61-note high-end digital/analogue hybrid synthesizer which offers 4x kinds of synthesis powered by 3x oscillators (Wavetable, Waveform, Particle and Resonator). The Fatar keyboard is velocity sensitive and features aftertouch and the expected Modulation and Pitch Bend wheels are present on the left of the keys.
SYN0006211-000

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TalkBack

Yang Tan

WILLIAM STOKES

Having cut her teeth as a studio engineer with a litany of global names, classically trained producer Yang Tan is moving into a new phase of her career. “I have had the pleasure of working with a lot of ‘top dawgs’ as an engineer,” she says, “but I don’t yet have the big names behind me as a producer. Having mega artists on my credit list is huge, but I want people to associate my name with a specific sound and style that is unique.”

Yang moved to Los Angeles after studying in Beijing. “I’m Chinese, born and raised. My initial plan was to come here and learn what I needed to learn, because I think that London, New York and Los Angeles are the three cities where you can learn the most in the music industry. There are actually great facilities in Beijing and Shanghai, but I think the whole music production process is not very well developed, so a lot of major artists will still come here to make their records, or to Korea or to New York. I was planning to study and work in the industry and then go back home. But I actually liked it here!”

It was Yang’s work at LA’s Paramount Recording Studios, assisting and engineering sessions and servicing consoles, that opened the door to bigger things. “From there I got a lot of opportunities and started working with a lot of big rap artists,” she says. “An interview with *Sound On Sound* is on my bucket list!”

At the moment I can’t stop listening to
Man, that’s a hard one. I can’t stop listening to Doja Cat. It’s mesmerising. I actually started listening to her years ago, before she blew up with ‘Say So’. The new album is really cool: sonically, the concept, everything. The character she pulls out is really cool. Oh, also The Kid LAROI, his latest record. He’s this Australian kid. He’s something special. He brings pop punk and rap into the same space and combines them really well. You hear it once and instantly remember it.

The project I’m most proud of
The next one! I always hope my next project is way better than what I’m working on right now. I’m always moving forward, always upgrading. That’s my hope. The next one will be what I’m most proud of, and I’ll keep going.

The first thing I look for in a studio
I always check the acoustic of the room. I go in and clap. And then I listen to records I’m familiar with on their system. That’s the first thing I check. That way I know what I’m working on. Some studios might have the best, most Instagram-ready look, but if the sound isn’t right you could be working on something very cool, and then go home and listen to it only to hear it doesn’t translate so well. So, the first thing I’ll do is actually go in the room and also get to know their system. If it’s lacking on the low end I’ll be careful on the low end. Or if the monitors are bright I’ll think “OK, their system is bright — I won’t be dampening that too much, otherwise I won’t have enough high end when I come out of the studio.”

The person I would consider my mentor
I have two. My professor at my university back in Beijing, the Communication University of China. His name was Jie Yang. He showed us the way to finding answers on our own and taught us how to think systematically. He really helped me. Even right now, if I have a technical issue I can’t solve, I’ll consult with him. He always has the right answer. Just from what I learned in his class I was able to design large-scale broadcasting systems while I was still in my second year of school. That’s when I got an opportunity working with an engineer at a large rock festival in China, called MIDI Festival. They were thinking about how to do the broadcasting and live recording, and I gave them the ideas, designed a system and also got them the sponsorship to get the consoles in. It was two main stages with over 64 signal channels at once, and with a dual backup at the same time. Jie Yang gave me a lot of really good advice, and I had him go over the system to improve it. That gave me the confidence to say “OK, I can do this.”

The other mentor... I want to say: J. Cole. His work ethic. There was this one situation where I was using Logic, still doubting myself in creating music, and he just came in and sat right next to me with his laptop, also producing music but with Ableton. We had a little talk, I told him how I was classically trained so I couldn’t really create anything without sheet music, and he said he just takes 20 minutes, noodling on the piano, trying things. That really sparked me. From then I just kept putting in the work every day, kind of growing out of my limited beliefs of myself. Looking back, I’ve grown so much. I’ve done so many things I would never have imagined I could do.

The biggest misconception about the role of the producer
I think the biggest misconception is that there’s a definite description of the job. A producer can be someone who just brings the vision together — they don’t necessarily need to know how to engineer or compose music — they just have to be the one to bring it all together. They might just have the ears for talent,

and the ability to build someone up from there.

My go-to reference track or album
I don't have one. Most of my projects are all over the place in genre and style, so I'll normally go to the top albums within a genre, or something that's similar to the song I'm working on, and listen to it for a hot minute before I actually jump into the project.

My desert island studio item
This question has got me anxious! I'll say a Neve 1073 preamp. With the Tube-Tech CL1B compressor. If I could have a 16-channel SSL 4000 G console that would be great... and maybe a 3600-Watt generator! I think the 1073 just has that colour, that fatness, especially for vocals. In the modern day that's mostly what we use it for, vocals. Also for bass it's amazing. I think the Neve 1073 is a one-for-all situation. It's the best product ever designed.

The piece of gear I'd bring back into production
The [*Roland Juno*] 60 would be a good one, the original. I think that it's just been used in like, every record I've ever listened to! It's the most seen and used piece of gear in the studio, but most of them are in poor condition now. Not many people actually own a 60. Sometimes the older gear... I don't know what it is, it just brings that extra something.

The producer I'd most like to work with
FKA Twigs! I've worked with her once, but not on her own music. She produces her own music and also works with other producers. I'd love to see her creative process. I've heard she never uses a metronome, they just jam. Each producer just has their own laptop with Ableton set up and they just jam, and don't even sync to a metronome. When I worked with her she was just invited into the session and we just recorded her vocal. That was it. But she was amazing through the whole thing.

The advice I'd give myself of 10 years ago
It would definitely be to believe in myself. To start doing what I'm doing right now, which is producing and just making music every day, no matter what project I'm doing, just putting in the time and work. And then planning a step-by-step schedule for making that music happen. Keep learning, practice, listening... and repeat day after day. If I started doing that back then I would probably be a very different person right now.

ZPlane Fennek

Audio Metering Plug-in

This BS.1770-compliant loudness meter offers some interesting and original features.

HUGH ROBJOHNS

ZPlane have recently launched a successor to their popular PPMulator metering suite. Instead of emulating obsolete but delightful mechanical PPMs, this new audio metering system, called Fennek, is rather more modern and it focuses strongly on measuring programme Loudness in all its attributes — it's aimed squarely at professional audio-for-video applications. Fennek combines multichannel bar-graph peak meters (supporting all formats up to 10 audio channels), with various numerical readouts, and a configurable loudness history chart. The GUI is customisable and resizable, and Fennek can be used either as a standalone and self-contained metering system or as a conventional plug-in (VST, VST3, AU, and AAX formats, with support for Windows 10 or Mac OS 10.15 and 11.5 for Intel and M1/ARM machines).

Overview

Naturally, Fennek incorporates the latest complete BS.1770-related Loudness metering algorithms and associated displays, and all the usual targets and parameters can be configured manually or called up from an extensive collection of built-in Presets, which align the meter to the published requirements of the world's broadcasters, Internet Streamers, and other popular audio outlets.

Momentary, Short-term, and Integrated loudness (LUFS) measurements are calculated and displayed (with variance ranges and maximum values for the Momentary and Short-term meters to assist in 'short-form' programme assessment), along with Loudness Range, Dialogue Integrated Loudness (and the percentage of dialogue detected in the audio material, per Dolby's Dialogue Detection algorithm), and True Peak levels.

So far so conventional, you'll be thinking, and you'd be right. But where Fennek stands out above its peers is in its handling of measurement history. Of course, most loudness meter plug-ins now include some form of history view, but Fennek's facilities are — as far as I know — unique in this area, and give this particular plug-in solution a very useful workflow advantage over its competition.

Most loudness metering plug-ins can start and stop their measurement process automatically when the DAW is started and stopped, thus ensuring the accuracy of the calculated Integrated

ZPlane Fennek

£111

PROS

- Comprehensive loudness metering suite conforming to the latest specifications.
- Resizable and customisable GUI.
- Unique ‘measurement session overdubbing’ feature via host timeline synchronisation.

CONS

- Windows platform development appears unfinished.

SUMMARY

A comprehensively specified and implemented loudness metering system in which the measurement data can be synchronised to a DAW’s timeline to allow sections of data to be updated to reflect mix changes without having to re-analyse the entire programme content.

Loudness value. While Fennek does that too, of course, it also offers a couple of additional ‘measurement time-base’ modes. The most basic and familiar of these is the ‘Elapsed’ mode, in which loudness measurements are logged with the digital equivalent of a stopwatch, without any absolute time reference. A second option is called ‘System’, and this relates the loudness logging measurements to the computer’s own system time — this, naturally, is normally the current time of day — so it is akin to logging loudness events against a clock on the wall, identifying precisely when any specific loudness event occurred. This could potentially be useful in real-time broadcasting, for example.

The Perfect Host?

The third, most innovative mode is called ‘Host’ and this locks the History log time-base to the DAW project’s own timeline. The benefit of this way of working is that there is always a fixed relationship between the audio arranged across the DAW’s timeline and the calculated Loudness measurements being displayed, and it’s this fixed correlation that allows something ZPlane call ‘measurement session overdubbing’...

What is ‘measurement session overdubbing’ I hear you ask? Well, imagine the scenario where you’re mixing the soundtrack for a long TV programme, and you discover at the end that several small sections exceeded the broadcaster’s loudness or true peak limits. Helpfully, Fennek’s History display reveals the precise time(s) where the measurement tolerances were exceeded — and in a way which is directly relatable to the DAW’s own timeline, so that it’s easy to go back to the identified positions and remix just the offending sections to ensure compliance.

Fennek allows you to scroll through the history log and not only obtain readings at any point in time, but also sync your DAW’s transport, so you can attend to problems the metering has indicated.

Some other metering systems might already facilitate that, too... but they would then typically require you to run the entire programme back through the metering software all over again to re-measure the Integrated Loudness and other relevant parameters. This is where Fennek’s ‘Host’ time-base option really does win out, because its ‘measurement session overdubbing’ completely negates any need for another full measurement pass — potentially saving considerable post-production time and cost.

“Where Fennek stands out above its peers is in its handling of measurement history.”

In this Host mode, Fennek’s history log data is synchronised to the DAW’s timeline, and by simply replaying across just the remixed section(s) the metering software captures revised measurement data for those sections which is automatically incorporated back into the measurement history obtained during the original pass, and a complete set of updated loudness measurements is instantly available for the whole programme. This innovation is more than just convenient, in many professional situations it could easily be a complete game-changer!

As with any new metering system, you’ll need to take a little familiarisation time before you can access all the displays at a glance. But I found Fennek to be a powerful and accurate Loudness meter that’s more up-to-date and comprehensively equipped than most, and easy to configure to meet any specific requirements.

I did experience a couple of crashes when used as a plug-in and in Standalone mode it resolutely refused to acknowledge any ASIO drivers, offering only to work with Windows Audio or DirectSound drivers. Reading the manual, it would seem there are still a few bugs to be ironed out of the Windows version, which I’m sure will be attended to soon, but it seems already to be fully stable in a Mac environment.

The Host time-base synchronisation function and associated ‘measurement session overdubbing’ capability stand out as the star features. Anyone involved in audio-for-video editing or mixing/dubbing could find that Fennek revolutionises the workflow, and that makes this a very attractive new metering solution indeed.

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Vocal Editing

Pitch & Time

Last month, we showed how to comp a great vocal performance. Now we look at how you can use editing tools to improve both timing and tuning.

MIKE SENIOR

In last month's mag, you'll find an article (https:/sosm.ag/dec12-vocal-comping) in which I offered some tips about how to comp a great vocal performance from multiple takes. Once you've done that, though, there may well be more editing work to do. There's been a lot of hand-wringing over the years about whether vocal pitch correction is 'a good thing', but it's so widely used in mainstream music styles that the listening public have for the most part come to expect recorded singers to sound superhuman in this respect. And while it gets less press, vocal timing is also comprehensively manipulated in most chart productions, because lead parts are often faded up in the mix to the point where they significantly impact on the song's overall groove.

As such, there's little avoiding corrective editing if you're trying to put out commercial-grade song productions. But achieving decent-sounding results can be far from straightforward, because it's easy to suck the life out of a vocal performance during the process. So in this article I'd like to provide some tips for tightening your vocal performances effectively without sacrificing their naturalness and humanity.

Performance Enhancement

You can make your life a lot easier just by preparing properly for the recording stage. After all, singers respond to what they're hearing, which means any inherent backing-track problem can make it tough for them to bring their 'A' game. So make an effort to complete corrective edits elsewhere in the production before the singer steps up the mic, and try to put together a sensible mix balance too. If the drums and percussion are too loud, for instance, it can be difficult to accurately judge pitch; too quiet and the rhythmic accuracy may suffer.

Some headphones don't have great bass fidelity, which can cause sub-heavy bass or kick-drum timbres to all but vanish from the foldback mix, thereby robbing the performer of vital pitching/timing reference points. Monitoring latency, however small, can also mess with a singer's pitching, which is why I usually prefer to avoid software monitoring if I can — either by setting up analogue monitoring or by asking the singer to slide one side of their headphones off so they can hear themselves acoustically. Some singers swear that adding vocal reverb to their headphones helps them pitch more accurately but I've never found that personally, so your mileage may vary!

Some singers have trouble pitching when listening to their voice through headphones. It can often reduce your editing workload if you switch off your DAW's software monitoring while recording, and ask the singer to listen to themselves acoustically by slipping one cup slightly off their ear (but not turned outwards!).

Minimising the capture of room reflections and background noise on your vocal mic can be another wise move. Pitch/time-processing algorithms find noisy and reverberant signals more challenging to analyse and process, and this basically translates into a greater likelihood that you'll end up with unpleasant processing artefacts. And one further recording tip for rhythmic songs: encourage the singer to move a little with the song's beat while they perform, since that can do wonders for the rhythmic accuracy.

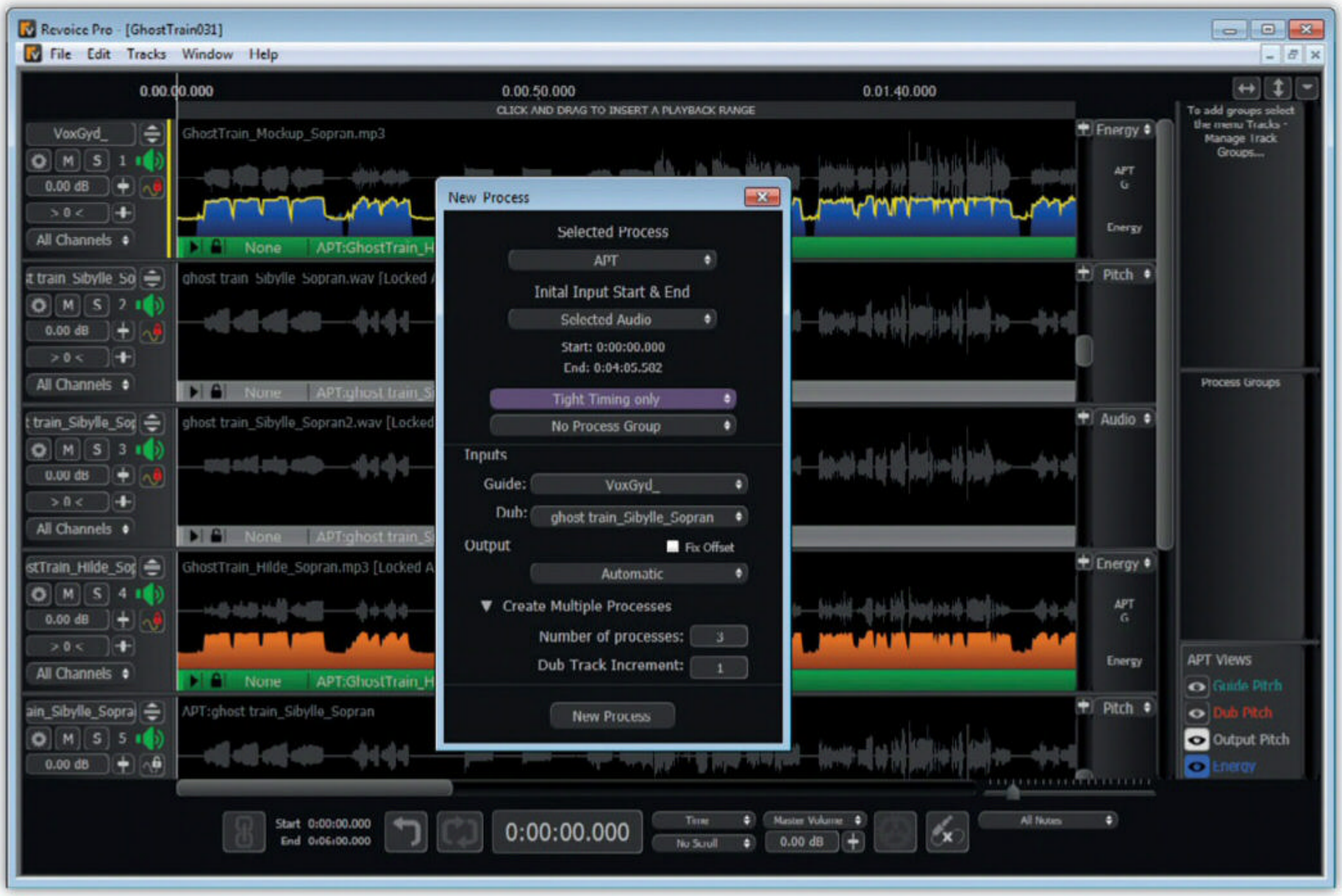
Pitch Perfect

There's a plethora of great pitch-correction software these days but, whatever you're using, and assuming you're not using it as a deliberate effect, the main principle to follow is to keep the processing to a minimum. The simpler and more targeted your pitch correction, the fewer unmusical side effects it'll produce. With this in mind, it makes sense to avoid set-and-forget automatic tuning-correction processes, because they process pretty much everything to some extent — irrespective of whether the music needs it or how much the processing is damaging the vocal sound. The fact is that most top-tier professionals still prefer to do lead-vocal editing work manually, so they can be guided at all times by their own ears.

I normally suggest working through the performance methodically, phrase by phrase, but then, within each phrase, correcting the most problematic notes first. What you'll discover by working this way is that many notes aren't actually that important to the impression that the singer is fundamentally 'in tune', and as long as you get the most important ones sorted out, any other pitching vagaries just add emotional colour to the performance. Trying to nail every little syllable to the pitch grid, on the other hand, is great way to kill a performance stone dead.

In a similar vein, try implementing your desired tuning via simple pitch offsets to whole words or syllables, rather than trying to iron out all the tiny variations within each syllable. Now, I realise that you sometimes encounter longer syllables during which the pitch wanders undesirably, and that some tuning software offers a pitch-variation »

Photo: wundervisuals



Batch-processing tools such as Synchro Arts Revoice Pro can be a timesaver when working with multiple tracks of layered backing vocals... but for individual lead vocals I find it almost always sounds better if I manually correct timing and tuning on a case-by-case basis, so I can judge each editing decision with my own ears.

» slider that seems to offer a quick fix for this. It's a slippery slope, though, because complacency with that slider can quickly lead to 'robotitis': a mechanical, synth-like vocal tone that's totally devoid of character. Much better to slice longer syllables into smaller sections for offset-based tuning.

With all that in mind, here are a few other specific tuning issues to bear in mind:

- Noisy components within vocal recordings (things like consonants and breaths) can cause problems for pitch-shifting algorithms in general, so you should avoid processing those if possible.
- If you notice that your pitch-correction software has misidentified the pitch of a note, take the time to remedy that manually. While this won't likely affect the pitch-shifting of that specific note much, it may well create smoother pitch transitions to and from adjacent notes.
- While most modern tuning correctors can tweak a note's pitch without altering the frequencies of its vowel resonances (called formants), I've found that pitch-shifts of more than a full tone can still make the vocal formants feel out of line. So don't be afraid to tweak the formant settings manually on those occasions.

Should I Start With Pitch Or Timing?

For years, I've always dealt with pitch editing first, followed by timing, although I mostly owe this habit to the practicalities of working with various versions of Celemony Melodyne (my own pitch-correction software of choice) over the years, and don't have a huge preference either way. Furthermore, I do quite a lot of comparative research of studio professionals, and there seems to be no broad consensus about where they start either. So I wouldn't waste too much energy worrying about it, and just start with whichever you prefer. Mind you, I wouldn't recommend trying to correct timing and tuning at the same time, personally, because I think each task feels to me like it uses very different 'muscles' mentally, and each is also demanding enough to require your full concentration.

A Stitch In Time

Many of the same software tools that offer tuning correction also provide audio-stretching facilities that can be used for timing-correction. Again, though, I'd warn against using any kind of bulk 'quantise' function for lead vocals, because that's never generated musical results for me. To be honest, I actually avoid time-stretching lead singers at all wherever possible, because I've never liked how that kind of processing affects the vocal timbre and reduces the sense of 'air'. I far prefer to use traditional audio editing techniques, which I demonstrated last month (*SOS* December 2021: https://sosm.ag/dec21-vocal-comping) and only resort to time-stretching as a last resort when my timing edits open up some gap in the line that's otherwise unbridgeable.

As with tuning, if you're working through a vocal performance phrase by phrase, try to resist the urge to pin »

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«The PRE-73 Jr is a great, affordable way into the classic 1073 sound, big and punchy – some might say larger than life, but with a musical heart. sound. The PRE-73 Jr is well worth auditioning.» «9/10»

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Matt Houghton, Sound on Sound , December 2018

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«The GAP PRE-73 and EQ-73 have become a staple in my studio. The colouration and sonic possibilities are very musical and stack extremely well. Excellent gear at any price, highly recommended!.»

«Fantastic gear! I've been tracking a little over the last few days, awesome.»

Malcolm Bruce

Additional Resources Online

To support this article, I've set up a special resources page on my website, where you'll find audio and video demonstrations of the editing techniques discussed in this article, downloadable multitracks with unedited vocal takes (so you can practice your corrective editing), and links to further reading.

W www.cambridge-mt.com/vocal-editing

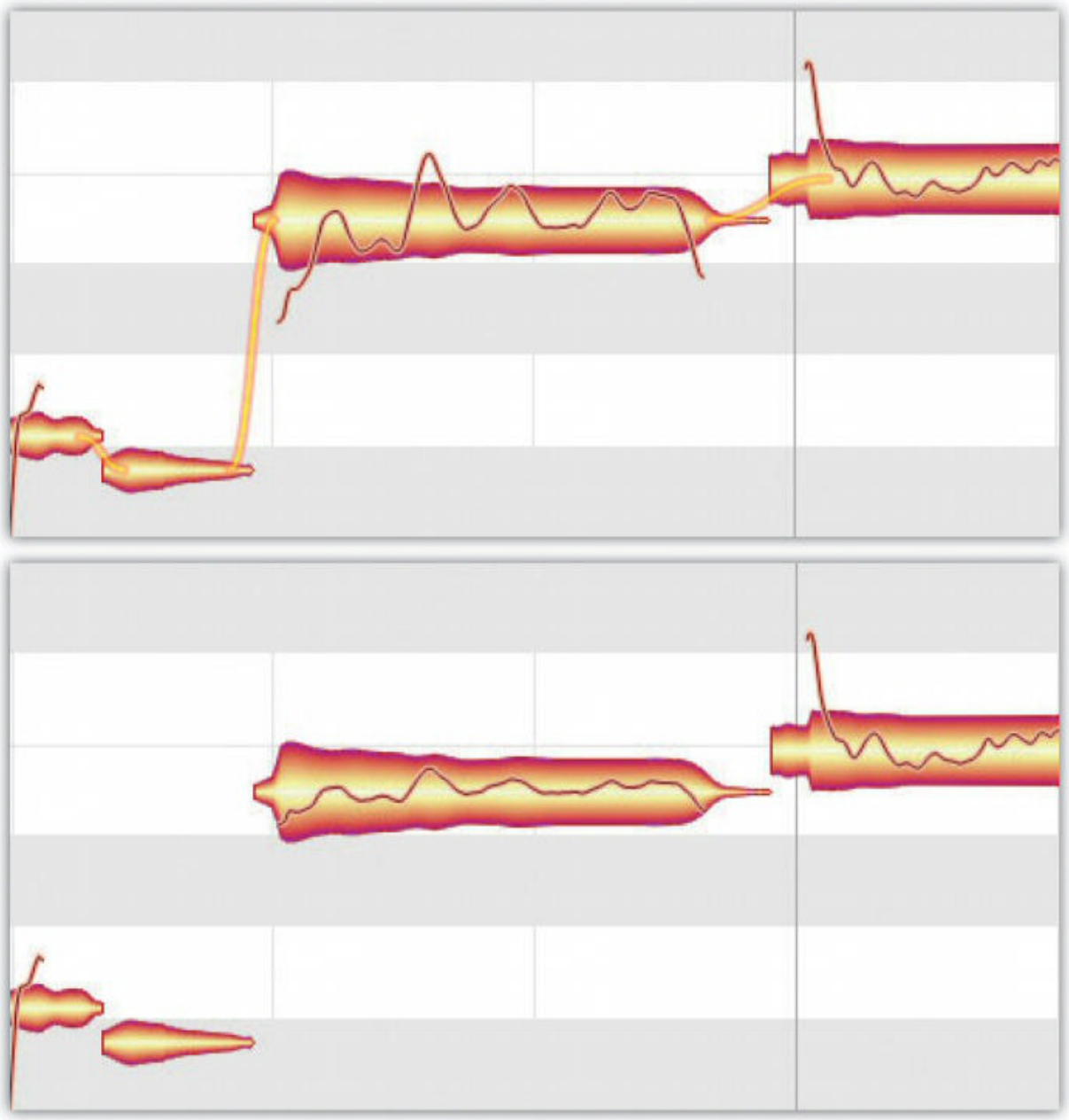
» every single moment to the grid. Focus instead on which syllables really have to be on the beat and which can afford to be looser without affecting the groove. Furthermore, I've found it a decent rule of thumb that the smaller the chunks of audio you're shuffling around to correct a perceived timing problem, the less natural the final edit will often sound.

The Eyes Have It

Whether you're editing tuning or timing, there are some important elements of psychology that you have to engage with. Most importantly, you must recognise that we humans are hard-wired to favour our eyesight over our hearing, and that this can very easily catch you out when working on any detailed studio task. If a vocal note 'looks' right on your DAW's pitch/time grid, your brain will strongly bias you to think that it sounds right whereas, in reality, a naturally in-tune and in-time performance will almost always deviate significantly from those grids. In the case of pitch, this may be because of small misinterpretations of the note's pitch within the software itself, whereas with timing issues there may be ambiguity in what actually defines the perceptual 'start' of a syllable.

For instance, if you look at the waveform of the word "crazy", you'll see that there's an opening transient for the "c", then a waveshape and level transition through the "r" sound before the more sustained "a" consonant fully arrives. Which of these elements constitutes the start of the note? Although you might instinctively put the "c" on the beat, this often (but not always) makes the word as a whole feel late, because the sustained "a" vowel then begins significantly after the beat. There's nothing cut-and-dried about questions like this, so the only solution is to continually challenge yourself to trust your ears more than your eyes.

Now, I'm not saying that your software's pitch and time grids can't be helpful in suggesting how a dodgy-sounding note might sensibly be adjusted — on the contrary, they can save you a lot of time. And in arrangements without a well-defined tempo grid, the waveforms of rhythm instruments can serve a similar purpose. However, you do have to discipline yourself to close your eyes or look away from the screen as much as you can while you work — both when choosing notes for adjustment, and when deciding whether your changes have improved the situation. For my own part, I also make a point after editing each song section (each verse, say) to get up from my studio chair and listen back from



The note in the middle of the top screenshot has been corrected to Melodyne's pitch grid, but it still sounds out of tune because of the pitch 'step' you can see within the note. Rather than trying to use the software's pitch-adjustment tools to simply iron out all the organic micro-level tuning variations (middle screen), the most natural-sounding solution to this is usually to separate the note into two sections, and then offset each independently as appropriate (bottom screen).

a different part of the room, facing away from my DAW screen! It's amazing how much clearer your decisions become once you step out of 'editing mode' like this and adopt more of a listener's mindset away from the controls.

Corrective Psychology

Remember that you'll be just as affected by the backing track you're hearing as the original singer was, so don't expect to be able to make appropriate tuning and timing judgements if the backing balance is nothing like it will be in the final mix. Similarly, for any mainstream work, it's vital to check your edits both on full-range monitors and under more bandwidth-restricted listening conditions, because differences in the audibility of the frequency extremes can make a big difference to how the vocal performance feels. I'd also recommend working at moderate volume levels, because pitch perception actually changes slightly with listening level, and in my experience, vocal tuning decisions made at higher listening volumes tend to skew slightly flat.

A lot of engineers don't consider the impact of their own physical movements while editing, either. I realise that it can be useful to bop along with the music to arrive at a clear opinion about where a vocal should sit within the track's groove — but there's also a sense in which this can be a bit self-fulfilling. In other words, if you're dancing around while trying to judge the success of an edit, your own physical sensations can overshadow the subtler rhythmic cues coming through your ears, and it's easy to fool yourself into thinking that your edit is more in time than it actually is. In fact, if I find myself wanting to move around while listening back to an edit, it often causes me to question the quality of the edit; I get suspicious that I'm trying to delude myself that it feels OK when it doesn't! (It's a bit like when you keep wanting to turn up the monitoring volume while mixing, which often serves as a red flag that the mix simply isn't good enough yet.) So I sometimes specifically do a final editing check without moving around at all, to be more certain of what I'm actually hearing.

Another psychological phenomenon you have to acknowledge is that you'll tend to get more sensitive to pitch and timing disparities the more you listen to any given phrase. Indeed, it's for this very reason that I think studio productions benefit from tighter tuning than live performances, because the listener is more likely to listen to the recording repeatedly. To an extent this works to your advantage while editing, because precise judgements get easier the more sensitive your ear becomes. I'll often hold off doing serious editing tweaks at the very start of any editing session, simply because I'm waiting for my ears to 'warm up' in this respect.

The flip side of this progressive increase in perceptual sensitivity is that it becomes increasingly easy to over-edit, simply because you start hearing pitch and timing nuances that will never really impinge on the consciousness of regular listeners. This is something I struggle with myself, as a bit of detail-oriented personality, so I have come to rely heavily on a few countermeasures. One is to avoid spending more than a few minutes on each phrase, concentrating just on making it better, rather than making it perfect. Then, after I've edited a whole verse, say, I'll listen back to it in its entirety and ask myself what actually still bothers me. As often as not, aspects of the vocal performance that seemed wayward when I was deep in the weeds end up just seeming desirably human once I've 'zoomed out' a bit. Moreover, I can have greater confidence that moments which still feel off-kilter do genuinely warrant more refinement.

> "In my experience, vocal tuning decisions made at higher listening volumes tend to skew slightly flat."

And, of course, you'll get even more objectivity if you do this kind of full-section playback after a cup of tea — in fact, in general, breaks are crucial when involved in such detailed work. For this reason, I've developed something of a rhythm while editing, where I'll work mostly in detail (with occasional full-section passes) for about 45 minutes, and then take a break. Tea and biscuits safely consumed, I'll play back the sections I was working on beforehand and make final adjustments to those, before starting the workflow cycle again with detailed editing of the next section.

Star Pitcher

With this advice in mind (and plenty of practice!) you should be well on the way to the ideal outcome of corrective editing — a performance that sounds like it hasn't been edited at all!

Backing Vocals

Although this article is primarily concerned with lead vocals, backing vocals often need corrective editing too. Here, however, I'm often happy to lift my moratorium against automatic tuning and timing functions, either because the backing part is low in the mix or because it's part of a layered texture — in both scenarios the audibility of undesirable processing artefacts is greatly reduced.

My main additional concern with backing vocals, though, is their harder consonants, especially when several parts are layered together, because flamming between consonants on different tracks not only sounds a bit scrappy (especially if the singers are panned in stereo), but can also impede lyric intelligibility. I also usually try to avoid backing-vocal consonants preceding those of the lead, because it can be quite distracting as a listener, especially over headphones.

Unmatched note-endings can also stick out like a sore thumb, so that's another thing to listen out for. When shortening an ending, I prefer not to just fade out mid-note, because this can sound rather unnatural. Instead, I usually slice a section out of the middle of the final syllable instead, and then close the gap by simply sliding the note's natural ending earlier in time (and using the matched-waveform editing technique described in last month's article about comping). When a note end needs lengthening, I'll tackle that in a similar way (ie. using simple audio edits) if there's any risk that chorusey time-stretching artefacts might become audible, but otherwise I'll frequently just spot-fix it with a bit of time-stretching.

Overall, it's as well to be aware that how tightly backing vocals are edited is often a strong genre marker, so before you get too carried away with minutiae, do check out a few relevant commercial releases to help you maintain your perspective.

How I Got THAT SOUND

Tom Syrowski

Incubus ‘In The Company Of Wolves’

JOE MATERA

LA-based producer and mixing engineer Tom Syrowski began his career as engineer for producer Brendan O’Brien, and has worked with Mastodon, Pearl Jam, Incubus, Bruce Springsteen and many other artists. Asked to recall his favourite sound, he identifies the vocal on Incubus’ 2011 track, ‘In The Company Of Wolves’.

“This is an epic two-part song that explodes in the middle, but it’s the vocal sound on this track that I am most excited about. We recorded this album at Blackbird Studio A in Nashville, which has an amazing and highly customised Neve 8078, originally built for Motown Studios in Los Angeles and later owned by Donald Fagen of Steely Dan. But one of the coolest things about Studio A is the unique airlock known as The Chamber, a small corridor that leads from the parking lot into the main tracking room where bands typically load in gear. When you stand in the space between these doors, about a 7x7 square, it creates an echo chamber. They went as far as building a motorised ceiling that allows you to adjust the reverb length of the chamber by simply flipping a switch on the wall, going from 8’3” to 18’11”.

“We had Brandon Boyd out in the live room singing on a Western Electric 630A, and placed a small powered PA speaker inside the chamber. We split the line, so that one line went into the control room where I was set up and the other line went to the PA speaker. We already had a pair of [*Neumann*] KM54s left up in the chamber from drum tracking part of the session and those, alongside the PA speaker, created a weird almost megaphone-sounding vocal. Lastly, I put a [*Shure*] 57 on the PA as a close mic. We weren’t sure at the time what the best blend was going to be, so I ended up recording all four of the mics for each vocal take — so when it came time to comp the vocal there was a lot of playlist-flipping going on. We ended using mostly the Western Electric 630 and stereo KM54s and never did use the close mic, as it ended up making Brandon sound like he was just singing through a guitar amp.”

A Song Of Two Halves

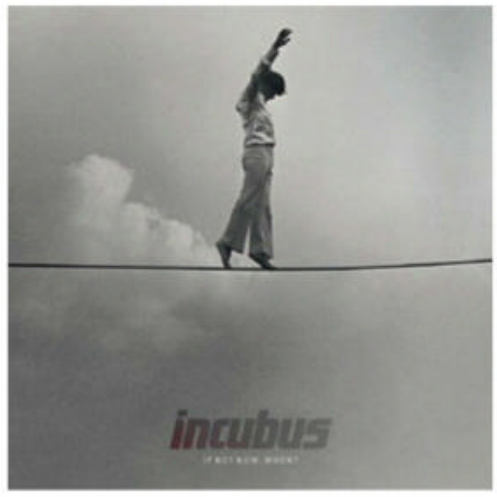
“The vocal on the first half of the song is the main sound, just the Western Electric 630A with the stereo KM54s blended in and Brandon singing in the live room next to the chamber. This part of the song is definitely more present and contains more of the main vocal mic, while the second half is much more effected. What we did there was put Brandon in the chamber with mics and the PA, and because of that, the vocal recording changed quite a bit. If you listen carefully, every once in a while, you can hear some nice feedback happening because of all the frequencies swirling around in the chamber and the PA blaring into Brandon’s microphone.

“It’s not just the vocal sound that changes from one part to the other. The first-half snare drum is very open and woody, almost like an old wood marching snare, and the kick drum is a wide-open Radio King, while in second half, the snare was switched to one of José Pasillas’ piccolo snare drums and the kick drum was swapped to a more modern Gretsch. All of this created a really cool dynamic I’m truly proud of”.

Hear The Sound

W https://open.spotify.com/track/2XCCNN8Z30n0DXsV2vZR8c

W www.youtube.com/watch?v=9Awfx69WbSs



No tricks
No copies
No gimmicks
No emulations
Only quality...

AUDIENT

You can't unhear quality

Reason Studios Reason 12 & Reason+

Music Production Software

There are now two ways to buy Reason, but whichever you choose, v12 is an essential upgrade.

SIMON SHERBOURNE

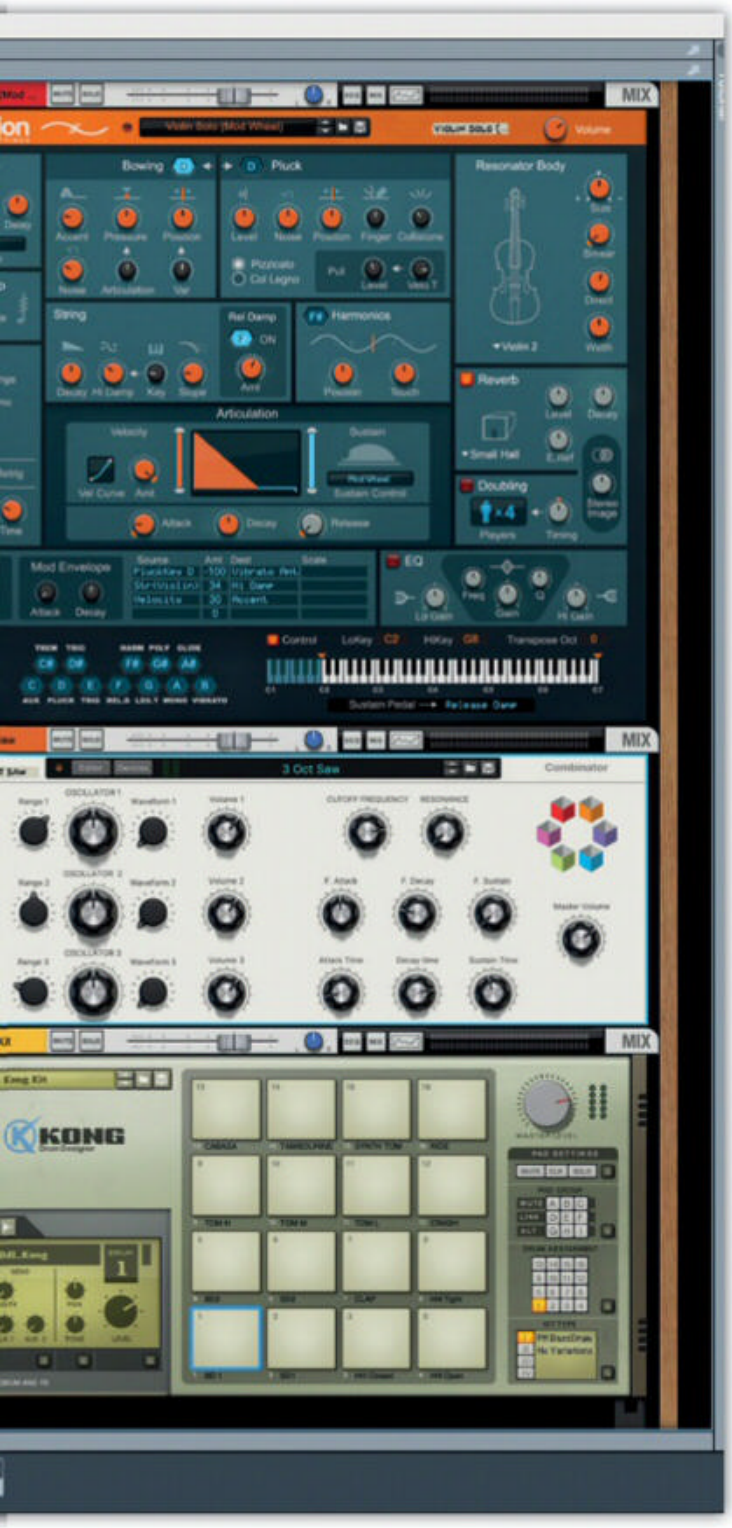
Version 11 was the start of a new double life for Reason, as the virtual Rack and its trove of devices jumped ship from the self-contained Reason production environment to set sail in other DAWs as a plug-in. This has been great for those of us Reason users who enjoy polyamorous relationships with our music software, and it was presumably an important milestone for the company too, who now have a much larger market they can address.

There have been some other changes on the business side since Propellerheads transmuted into Reason Studios, with the company embracing a subscription route under the banner of Reason+. Having cleared the decks and set a new course Reason Studios are focused on what they do best: making cool fun stuff to produce music with. And I promise that's the end of the maritime metaphors.

The prospect of subscription may have split opinions among Reasonistas (see the Reason+ box), but nobody can take exception to the choice of new features that have appeared in Reason 12. Reason Studios have tackled what must have been the three most popular and long-standing feature requests: high-resolution graphics, a more powerful Combinator and a simpler, more playable sampler.

Clear Vision

Reason has always had a highly visual focus. Much of its popularity stems from



The updated graphics engine brings the wow factor back to the Rack.

Mimic: an inspiring new device that brings fast slicing and sample warping.

the revolutionary (at the time) virtual rack of skeuomorphic instruments and devices. Everyone remembers where they were when they first saw the Rack spin around to reveal those wobbly patch cables. But since Reason was launched, computer screens have gained steadily in resolution, leaving the fixed-width Rack looking a bit on the poky side on a typical HD monitor, and grainy on today's high pixel density 'retina' displays.

The programming guidelines for Reason's devices and Rack Extensions have actually been preparing for this for several years, requiring high-resolution graphic assets for the panel interfaces. But the Rack itself was stuck in the noughties and couldn't take advantage of these resources. With Reason 12, a mammoth overhaul of the graphics engine has brought things up to date.

The Rack, Browser and Mixer now have native support for the double resolution offered by retina, UHD or 4K displays. In other words, for a given size on screen, a device will appear significantly sharper and more real. The effect is pretty startling, aided by the fact that the panels have been reworked with improved detail and lighting.

The whole application is also now scalable, meaning that the size your Rack appears on screen is no longer tied to the global screen resolution/scale you've set in your system. This is really useful whether or not you have a UHD display. What you can't do, unfortunately, is scale the Rack independently from the rest of the app. I guess the intention was that scaling would be a set-and-forget option simply to balance the overall scale of Reason against your OS, but a lot of early user feedback agrees that we'd love to be able zoom in on the Rack itself when working with devices. Reason Studios have taken this on board (sorry), so hopefully we'll see this in the future. This is of course not an issue for the plug-in, which, being just the Rack, already scales independently from the DAW it finds itself in.

Sampling In The Age Of Reason

There are several samplers in Reason, most of which date back many versions. The classic NN-19 and NN-XT devices have been the workhorse sample players, primarily used for powering »

Reason Studios
Reason 12 & Reason+
£499

PROS

- Rack facelift for modern displays.
- Custom Combinator Panels.
- Time warping, chopping and vocal manipulation on the new sampler.

CONS

- No independent Rack zoom.

SUMMARY

It feels like a new era for Reason with a makeover both on and off screen.

Reason+ sound packs are already taking advantage of the Combinator macro possibilities.

You can now add and edit your own Combinator controls, as well as change the size, colour and backdrop of the panel.

» multisampled instruments in the factory library and third-party sound packs and refills. Then there's ReDrum and Kong to take care of sample-based drum kits. Reason has not had a need for a bells and whistles modular sampler like Kontakt, as you could achieve similar results by patching devices together in the Combinator.

However, while the NN units resemble S-series Akai samplers, and Kong follows the classic MPC layout, they are missing some of the key features that can make sampling a fun and creative musical technique. They are good for programming and making patches, but you can't just drop a sample into them, chop it, stretch it and play it with immediate results.

Reason 12 introduces Mimic, a new sampler instrument that focuses on quick results, experimentation and performance. In its simplest, default mode (Pitch) you can drop in a sample (or record in directly) and immediately play it back from the keyboard. In this respect, and also in the general appearance of the panel, Mimic has similarities to the Grain instrument which appeared in Reason 10, with a large waveform display and set of modules below for settings, modulation and sound shaping. However, it's been kept simpler than many of the flagship Reason instruments, without a modulation matrix or envelope designer.

Slot Machine

Mimic has eight sample slots, all of which have independent panel settings. Depending on the mode, the slots are used differently. In Pitch or Slice Mode, only the selected slot is active at any one time. Multi Slot mode lays out the samples across the keyboard, repeated every octave. This is ideal for using Mimic to trigger a kit of drum sounds or other one-shots. Multi Pitch lets you map slots to keyzones like a traditional sampler.

Slice mode fills one of the most obviously gaps in the Reason sampling arsenal. Although Reason has been equipped to play back pre-sliced samples with the Dr REX instrument and Kong, until now if you wanted to chop a sample you needed to do it in an audio track in the sequencer. Mimic brings Reason up to speed with other performance samplers by building slicing into the device itself. By default slice points are dropped automatically by transient detection, with the usual Sensitivity knob for manual assistance. You can also freely add, delete and move markers, but you can't do a manual real-time chop as you might on many hardware workstations.

While slicing was top of my Wishlist for the new sampler, it's been the time-warping powers that have really inspired me in Mimic. You get independent control of pitch and playback speed using various stretch modes. In the default Tape mode Mimic behaves like a traditional sampler, pitching notes across MIDI notes by changing playback speed. In other modes, playback speed is consistent across keys, with various stretching schemes used for repitching.

This is not revolutionary stuff, but there are two things that really stand out. First is the exceptional quality and range of Reason's Advanced stretching when applied here. You can slow down the playback speed all the way to zero (actually a tiny loop) and get incredibly smooth results. You can modulate the Speed around zero and get granular style scanning, except that it sounds much better than any granular! The range of textures you can get out of a single sample is amazing.

The other addictive mode is Vocal. On the face of it this is a stretch mode »

Modern Classic.

The 1608-II Recording Console from API

RPM Music Group,
Rancho Cordova, CA

Red Barn Studio, Springdale, AR

NHK Technologies, Tokyo, Japan

SM Entertainment
Seoul, South Korea

CT Studios, Chesterfield, VA

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Cleveland, OH

With a powerful feature set and API's optional cross platform ***Final Touch Automation***® system, the 1608-II continues API's rich legacy of crafting the world's highest quality analog recording consoles.

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SX Pro UK: www.sxpro.co.uk, + 44 14 6241 4196, sales@sxpro.co.uk

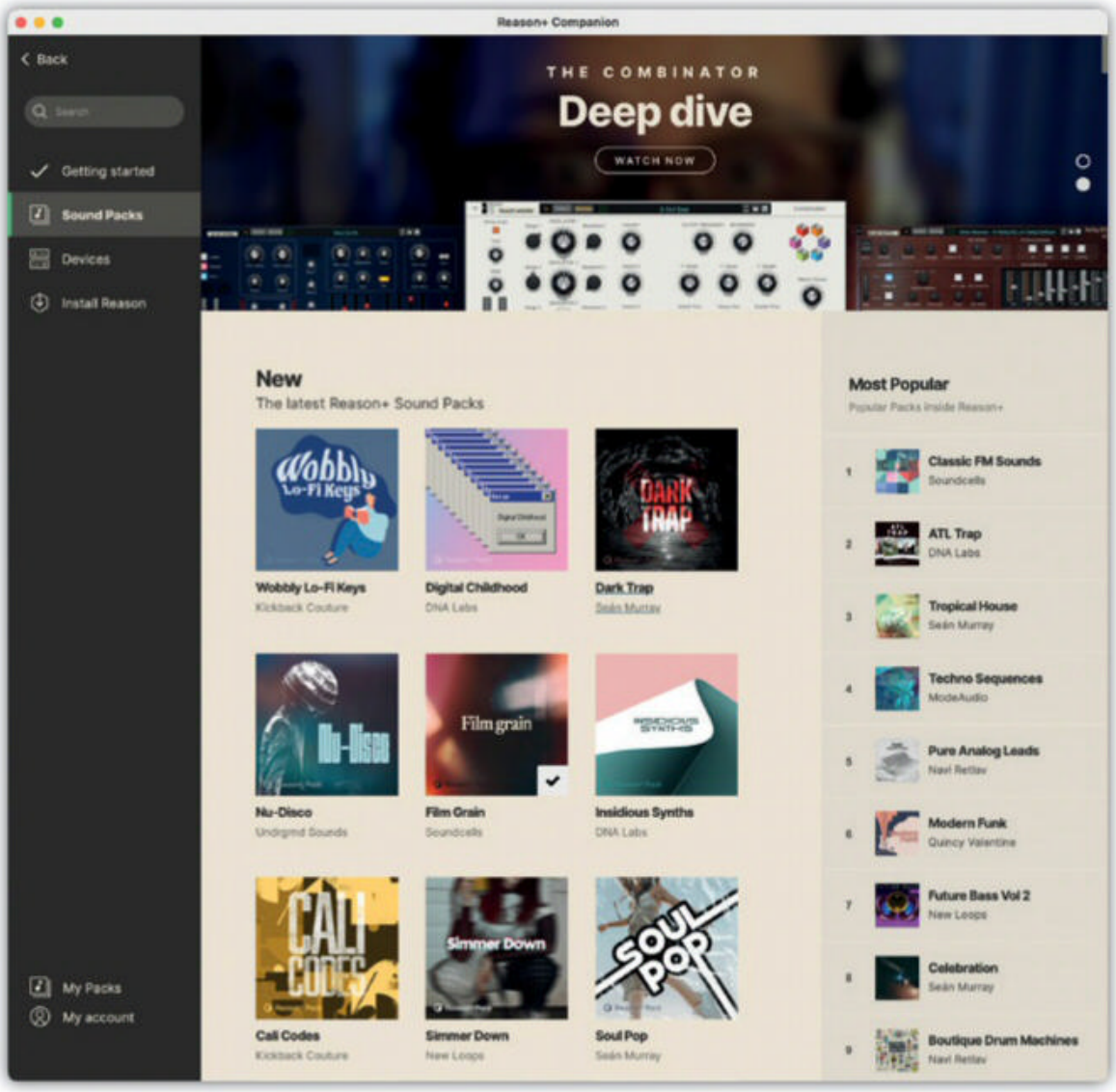
Reason+

There are now two ways to use Reason. You can go the traditional route: buy it and own it, pay for upgrades when you like, and purchase additional Reason Studios premium instruments if you want them. Or you can opt for Reason+, a subscription service that gives you access to everything while you stay on the plan. The Suite option, which was the previous way to buy the whole collection, is now discontinued.

If you judge reality by YouTube comments and social media, then subscription is something that software companies love and users hate. It's certainly true that subscription is great for a company: it provides a more predictable, regular income than a sporadic upgrade cycle, and investors in particular love predictability. But it's also popular with new users, who get to start using the software without a large initial outlay. The total cost over time is likely to be higher, which is why the pot gets sweetened by including extras like instruments and sound packs. This of course might sting if you're a long-time user who's already bought the software and some extras.

Overall I think Reason Studios have managed to steer a careful course through these waters. They've kept the original buy-to-own model for those who don't want anything to do with subscription (other than dropping Suite), and they also have a "cancel any time" policy so you're not committed if your circumstances change, and you can drop in and out if you're a casual, occasional user. The only thing that remains to be seen is whether the price is right.

So, what do you get if you join the Plus Club? As well as the Reason app and plug-in, you have full use of everything Reason Studios makes, bar ReCycle. That includes all the awesome recent instruments like Complex-1, Friction, Algoritm, and earlier must-haves like Parsec. There's also the premium Players like Drum Sequencer, PolyStep Sequencer and Pattern Mutator.



The Reason+ Companion app manages your devices and sound packs in one place.

An extra that only comes with Reason+ is access to Sound Packs. These are small collections of 10 patches with a particular theme, created by various producers. They seem to be aimed at providing inspiration boosts. The collection is added to weekly, and is accessed via a new app called Reason+ Companion, which also manages updates and installation of devices and Rack Extensions. The Packs are a nice idea, and the collection has already grown to include some quality stuff. Once installed by Companion, they appear in the browser in a dedicated Packs category. Each pack is a folder which you have to dig a couple of levels into to find the patches, and this could do with tidying up.

» with a movable formant control suitable for good results with vocals. But it's the Fixed Pitch mode that steals the show. In this mode, the sample is forced, auto-tune-style, to the pitch of the MIDI note you play. I've got slightly obsessed with a trick where I load in a whole line of vocals, switch the Start position to Global (which keeps sample playback position continuous as you trigger new notes), and choose the Fixed Pitch vocal mode. You can then replay the phrase with any melody, or simply play the original melody for a unique take on the Auto-Tune sound that continues to permeate so much modern music production.

Combine & Conquer

The Combinator device was probably the most important addition to Reason since its launch. It provides a way to pack a multi-device patch of other rack units into a self-contained sub-rack which can then be treated like a single chunk. The composite device has its own audio, MIDI and CV connectivity, and it has a macro panel for controlling selected parameters within. Similar concepts are now in other DAWs, for example Live's Instrument Racks or Bitwig's Containers.

Combinator has many practical and fun uses in Reason. I've used it for live performance, setting up a different Combi for each song with samples and keyboard splits pre-arranged, along with any performance Macros I need. They're great fun for inventing new instruments and effects from combinations or chains of other devices. They also provide an easy way to transfer something you're working on in the standalone Reason to the Reason Rack plug-in.

Combinators have become the main way to create preset patches in both the Reason factory libraries and third-party Refills and sound packs. As a Refill developer, I moved from creating patches for individual devices to Combinators as soon as the device was launched. The format provides more scope for recreating the authentic sound and control of an instrument you've sampled, or for coming up with completely new device ideas.

Inevitably, though, these sound-design projects would hit inherent limits in the Combinator environment, in particular the fixed macro control provision of four knobs and four buttons. There was also a limit to how many mappings you could make to each device. Reason 12

This Combi from www.randomnoisemusic.com uses a custom backdrop and macros to create a new instrument.

introduces an overhauled Combinator. We're not just talking about a few extra controls, you can now completely customise the macro panel.

A new Configure mode in the Combinator editor lets you add control elements from a library of different knobs, buttons, sliders and faders, and position them freely on the panel. (Some handy tools help you align and space the elements you create). Apparently the library will continue to expand with different graphical styles. In addition to this, you can now adjust the size of the Combinator panel. It was previously fixed at 2U, but can now range from 1U to 6U. The background colour can be changed, and, as before, you can import a custom background image, which now supports the high-resolution graphics that came in with Reason 12.

Reason+ sound packs and the built-in Reason Sounds collection are already beginning to do great things with the new Combinator, as you can see from some of the screenshots here. Reason Studios are also talking about further developments that could make the Combinator even more of a sound designer's dream.

Conclusion

With this launch Reason Studios have shown that they can still deliver a traditional big box upgrade release, ticking off three big items from the very top of the wishlist, and bringing important modernisations to the underlying tech. At the same time they've been canny in demonstrating the benefits of a Reason+ subscription: early access to feature releases, sound packs that take advantage of the new devices, and of course the keys to the whole toyshop of Rack Extensions and Players. Either way it's an essential update for existing users; and for anyone looking for a huge modular plug-in suite of instruments and effects Reason is more desirable than it's ever been.

£ Reason 12 £499, upgrade pricing from £199. Prices include VAT.
W www.reasonstudios.com

Slate Digital MO-TT

Multiband Upwards & Downwards Compressor

Slate have taken the famous OTT effect and added a whole load more control, creating a unique dynamics processor.

MATT HOUGHTON

The OTT preset for Ableton Live's Multiband Dynamics plug-in has been used and abused for years, not least by EDM producers, to breathe new life into boring or static sounds. So popular has it been, in fact, that Xfer Records developed a freeware VST/AU/AAX plug-in to make it available to users of other DAWs. Now, with MO-TT, Slate Digital have picked up the OTT baton and run the next leg, adding various features that should extend its appeal way beyond OTT's traditional user base.

MO-TT is available in the usual formats for Mac and Windows operating systems. Like all Slate's plug-ins, it can be purchased separately, and is also included in their All Access Pass subscription service. Authorisation uses iLok, though you don't need a physical dongle, and I found installation to be quick and hassle-free.

The Ups & Downs Of OTT

Some readers may find it helpful if I start by spelling out what the original OTT does, and the appeal it holds. In essence, OTT is a two-stage, three-band compressor. The first stage performs upwards compression while the second applies downward compression, both using the same crossover filter settings so that they operate on the same bands. Like conventional downwards compression, upwards compression reduces a signal's dynamic range, but it does so by lifting low-level signals up towards the threshold. In a sense, it's similar to aggressive parallel compression — that too can be used to raise low-level details — but with upward compression the 'lift' of low-level information is modulated according to the attack and release times, the curves and the ratio.

Each of OTT's bands has two threshold controls, one for each stage. You can tweak the crossover frequencies, the time constants and the ratio in each band. You can also change the overall gain for each band, 'scale' the attack and release times across all bands, and there are also per-band and global controls for scaling the amount of compression. There are a few more settings but, taken together, those I've mentioned form a quick and convenient means of adapting presets to work with pretty much any source.

While the exact nature of the resulting compression depends on the settings, the combination of 'push and pull' means OTT can (though doesn't necessarily) heavily restrict a signal's dynamic range, with the levels in each band often occupying a sort of 'middle ground' between the two thresholds. The variable phase shifts and cancellations around the crossover frequencies also make significant contributions to the sound — the adjacent bands always interact where they overlap. Typically, the HF band raises any hiss and air too, making the overall sound brighter, and it tends to modulate those sounds. (Though, of course, you can reduce the amount of compression and/or level of that band, it you don't want that brightness.)

The main appeal is that the result is typically anything but 'middle ground'! Get it right, and everything seems to pump, breathe and come to life. Movement in the source — the quick, rhythmic level changes in a drum loop, say, the cyclical swooshes of modulation effects and filter sweeps, or a ping-ponging stereo delay — tends to be emphasised, too. It's not a tool for every job, but hopefully you can see why it might appeal!

Slate To The Party

Now, you might be wondering if you could achieve all this simply by combining your existing plug-ins. You might: I managed something similar with two instances of Fabfilter's Pro-MB, but I found it tedious and time-consuming to set up, and trickier still to adapt the settings for use on different material. Slate's MO-TT, on the other hand, grants you the authentic OTT experience right off the bat. For all the material I tried it on, its default setting sounded pretty

The Advanced view makes many more parameters available, which should please the tweakers amongst us!

much identical to OTT on its default setting. MO-TT is equally easy to adapt, too, yet it also offers way more control and subtlety should you desire it.

MO-TT's scalable GUI first opens in Easy view, intended for those who want, simply and swiftly, to adapt the presets. I'm sure plenty of users will be very happy to remain in this screen, which has three Quick Set presets (the classic OTT setting, Hip Hop and Vox) and a pop-out folder giving access to many more, with subfolders for different material: Master, Instruments, Vocals, Drums and FX. Most presets sound pretty aggressive, but that's actually useful, as their 'assertiveness' reveals their character instantly. Once you find one with a suitable action, you can simply tweak the input level control, then scale the global compression Amount and Timing and, if you need, the individual band Amounts and levels, and then set the desired output level. That's about 90 percent of your work done...

Slate Digital MO-TT

$149

PROS

- Nails the default OTT sound.
- Easy view makes it intuitive and accessible.
- Advanced view makes it more versatile.
- Good range of presets, which are easy to tweak.

CONS

- None.

SUMMARY

Slate have made the classic OTT multiband two-stage compressor effect more useful to people working in a wider range of genres.

Much of the remaining 10 percent can also be done in Easy view. You have full-band input and output meters and level controls, switchable multiband clipping, the (global) option of soft or hard-knee compression, and a choice of peak or RMS sensing. A soft knee and RMS sensing generally sound smoother, while a hard knee and peak sensing make the effect more aggressive and in-your-face. Generous space is given to the compression metering, with a ±36dB scale, and a grabbable white handle on each band's meter for a level control. These are joined by two Tone controls (a circa 150Hz LF shelving cut or 50Hz bell boost, plus a more complex bell-based boost/cut on the top end), global Amount and Timing controls to scale the effect, and similar Amount controls for each band.

Had Slate stopped there, this would already be very a handy plug-in. However, Advanced view gives you more. You can use the main metering/crossover graphic to boost and cut each band, or change the crossover frequencies. Do the latter and you'll realise that this can actually be a five-band device, with the three bands of compression flanked by unprocessed highs and lows. This may sound trivial but I found it surprisingly useful: with the aid of the inbuilt frequency analyser if you need it, you can easily focus the three bands on the areas in which all the interesting tonality resides, while avoiding lifting any problem sounds above or below them.

You're also treated to global and per-band controls for the input level and the thresholds for both compression stages, along with bypass and solo controls for each band. A global filter slope control, with three options, has a far greater influence on the sound than you'd imagine, with gentler slopes causing more overlap and thus greater interaction between the bands. Side-chain options become available too: you can choose between external and internal, and whether bands should respond to the full signal or only in-band frequencies; there's also a variable stereo link, again both global and per-band. Two global threshold controls and a pair of attack/release sliders allow you to increase or decrease the current settings on all bands, while you also get three Timing Style curves: Classic, Smack and Smooth.

Test Spec

I used two computers to test MO-TT in Cubase Pro 11 and Reaper 6: a Scan Windows 10 PC with an Intel Core i9 9900K processor clocked to 4.9GHz and a 2018 MacBook Pro Retina running Mac OS 10.14.1.

Verdict

In short, you have a hell of a lot of control here if you want it — way more than is offered by the original OTT. And it's done in a really intuitive way, with Easy view keeping everything simple and accessible, and Advanced view delivering more options for those who like to refine things.

So, while MO-TT can certainly 'do OTT', it's capable of far more besides, whether you're aiming for subtle treatments or more out-there effects. It's a serious option for those working in, say, pop, rock or R&B. For instance, I found a subtle touch of MO-TT great to revoice problem sounds in a couple of rock mixes — I revived flabby kicks and boxy snare drums and, notably, coaxed some life and brightness out of a dull-sounding rock vocal part that had been captured using a dynamic stage mic. It's not just a mixing tool, though. When writing some ambient electro and pop tracks, I found it a wonderful means of breathing new life into my DAW's loop library, and for injecting a sense of movement into synth pads and delay tails. Sound designers could love it.

Like OTT, MO-TT won't be everyone's cup of tea — but it should bring the same smile to the faces of a much wider range of musicians and producers. I was surprised at just how versatile this effect can be, and I won't be removing this one from my plug-in folder any time soon.

£ $149. Also available in All Access Pass, from $9.99 pcm.
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Akai MPC Studio

Sampling Workstation

The MPC Studio provides an affordable middle ground between Akai's software and hardware MPCs.

SIMON SHERBOURNE

The MPC has reclaimed a lot of lost ground in the last four years, thanks to the excellent standalone workstations, MPC X and Live. More recently, a very capable free version of the MPC software, dubbed MPC Beats, has brought new users to the MPC platform. This may have been an effective gateway drug strategy: if you've been hooked into MPC as a plug-in, the natural next step would be to add a dedicated hardware controller.

Studio is Akai's next-generation hybrid MPC, a pure controller that acts as a hardware front end to the MPC desktop software and plug-in. MPC and MPC Beats actually have generously open MIDI control mapping capabilities, and factory templates for other popular controllers, but it's always hard to match the integration of a purpose-designed tactile surface with control sets, workflows and visual feedback aligned with software.

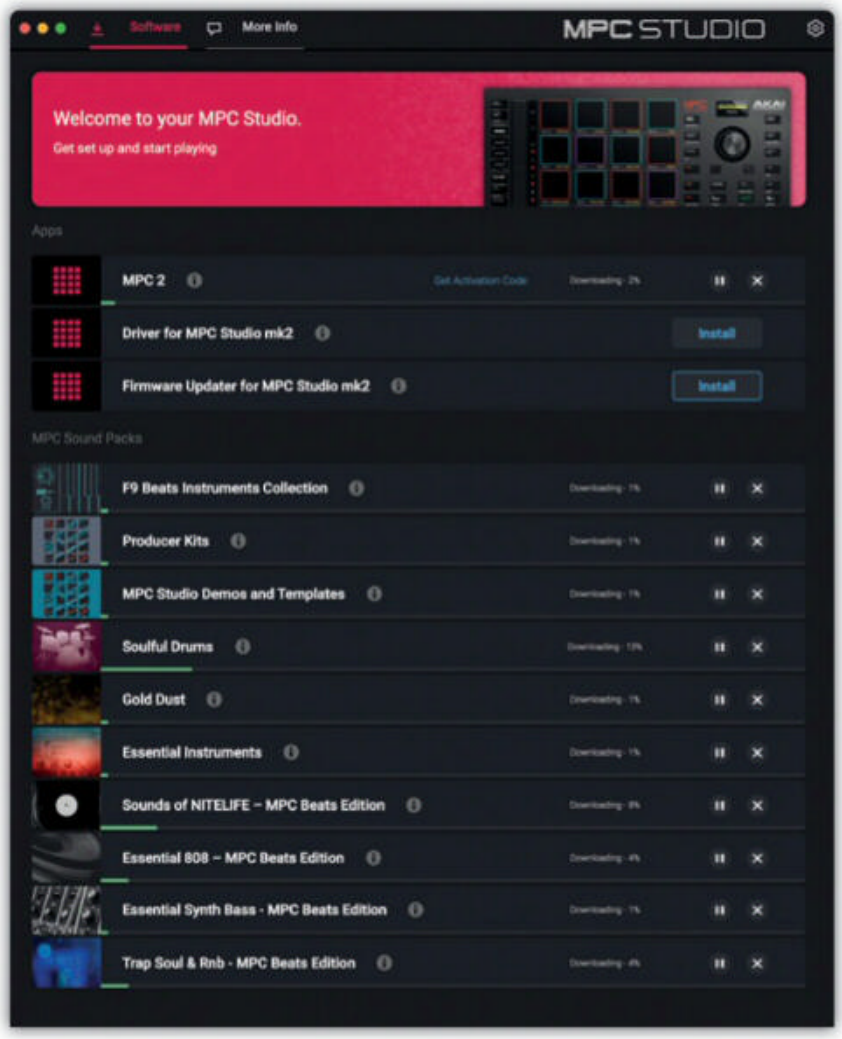
Unboxing

My first impressions of the MPC Studio were really positive, it's an attractive and low-profile slab, it feels really good and seems solid and well built. It has my favourite set of physical properties for a hybrid controller: it slots tidily between a keyboard and computer display, is powered by USB, and fits in a bag. It made me excited to get stuck into playing it.

The MPC Studio comes as a combined hardware and software package, with the full version of the MPC software, a library of kits, instruments, loops and samples and the complete suite of virtual instruments you get with any other MPC. In fact there are a number of tasty new instruments and effects since the last time we reviewed an MPC: see the 'Force 10' box. The hardware will automatically spring into life any time the MPC app or plug-in is running on your computer. The Studio relies on your computer system and other peripherals for audio I/O, but it does act as a single-port MIDI interface via a pair of mini-jack connectors.

First Steps

To recap what MPC has to offer, it's a multitrack workstation offering drum kits, sampled and synth instruments, loop launching and sample chopping. It's also a MIDI sequencer, VST plug-in host and audio recorder! Projects are structured around multitrack song sections called Sequences, which roughly equate to Patterns on a traditional drum machine, but without the length restriction. This is all backed up by a modern, flexible mixing environment that makes MPC a DAW in its own right, even if you only use it as a drum machine plug-in.



The Software Manager gets all your ducks in a row, including a nice collection of sound packs.

Although the MPC Studio is not trying to be an immersive hardware experience in the same way as the standalone MPCs, I thought I'd see how far I could get before I reached for the mouse. The display is small, but it's high resolution and colourful and manages to pack a reasonable amount of information in. I could see I was in Track Select mode and the big data wheel moved me through MIDI tracks. The Browse button is front and centre, and tapping that put me into the Media Browser, showing a scrollable list of all Drum and Keygroup Programs in my library. I discovered that pushing Browser again toggles to a file browser for loading individual samples.

Like all MPCs, the Main button got me back to the default position, and it was pretty easy to figure out that Program Select would let me assign the kit I'd just loaded. Having done that the pads lit up and I could get finger drumming. The pads feel the same as the current standalone MPCs, which is to say excellent, and the whole unit is very stable on the desk when played. Note Repeat of course gets a dedicated button, with repeat rate controlled from the touch strip — a nice innovation compared to the standalone MPCs.

Holding the Tap Tempo button let me set a tempo with the encoder. Next I wanted to set the length of my first sequence, and enable the metronome, but at this point I had to look to the software, as there doesn't seem to be a way to do either of these things from the controller. As you might expect, without the large touchscreen interface of its bigger siblings, the MPC Studio primarily focuses on all things related to performance, with settings and options accessed on the computer.

Muscle Memory

The transport controls are laid out like on other MPCs, and it's easy to record and build up a beat, overdubbing on to a single drum program track. Browsing the rest of the control layout is kind of interesting, perhaps revealing what Akai believe are the priority controls for a modern MPC user. There's now a Quantize button, giving you direct access to post-record quantisation in addition to the traditional Time Correction toggle. Pad Mute/Track Mute modes get a dedicated panel button now, which I guess recognises the fact that many people work by building up patterns »

Akai MPC Studio
£229

PROS
- Great pads and overall build.
- Compact and USB powered.
- Touch FX.

CONS
- No Q-Link Encoders.
- Layout and Mode shortcuts inconsistent with other MPCs.

SUMMARY
A very desirable little hybrid MPC and desktop controller.

MPC 2.10 adds new instruments and effects, like the Mellotron and AIR Stutter.

» in a single program and bring channels in and out, more often than arranging with multiple Sequences.

Frustratingly there's little consistency at all between any of the current MPCs with regards to what operations and modes get buttons, and where those buttons go. The Studio does away with Mode buttons altogether apart from Main and Pad Mute. Instead Modes are selected with a modifier and the drum pads. The three standalone MPCs share a shortcut system for accessing modes from the pads, but the MPC Studio assignments are different, which just seems crazy. My muscle memory for jumping to Prog Edit and Sample Edit views now takes me to the Sampler and Looper instead.

For sample editing the MPC Studio provides three dedicated buttons that switch the encoder between Sample Select, Start and End. Control resolution defaults to coarse, but holding Shift allows for fine adjustments. Chopping a sample is handled in the usual way from the pads, with slices edited like any other samples, but if you want to process a sample, assign it, or convert slices to a new kit you'll need to take care of that from the software side.

Touchy Subject

The most notable things missing on the MPC Studio panel are encoders, or Q-Links as Akai call them. All other MPCs have four or 16 knobs that can be used for controlling and automating plug-in and mixer parameters, or for things like sample editing. Even the original MPC Studio had four (as well as a rather more functional screen interface).

On the MPC Live, I tend to use the main encoder more than the Q-Links, although that's because it's easy to target a parameter on the screen. I definitely missed having the bank of knobs for controlling instruments and effects, and for sample editing.

Instead of encoders, the MPC Studio offers a Touch Strip. A mode selector switches the functionality of the Strip away from its default Repeat Rate to other functions like Pad or Program level, Pitch and Mod Wheel. A Notes mode lets you play notes from the strip, and you can define a root, scale, and range. It could be improved by allowing you to hold notes on the pads to strum, as you can do on the Arturia Keystep 37 or NI Maschine.

The strip can also stand in for a single Q-Link. MPC always manages Q-Link assignments in chunks of 16, either pinned to a particular function, or fluidly following the current mode or view. Choosing which of the 16 the Touch Strip is controlling is

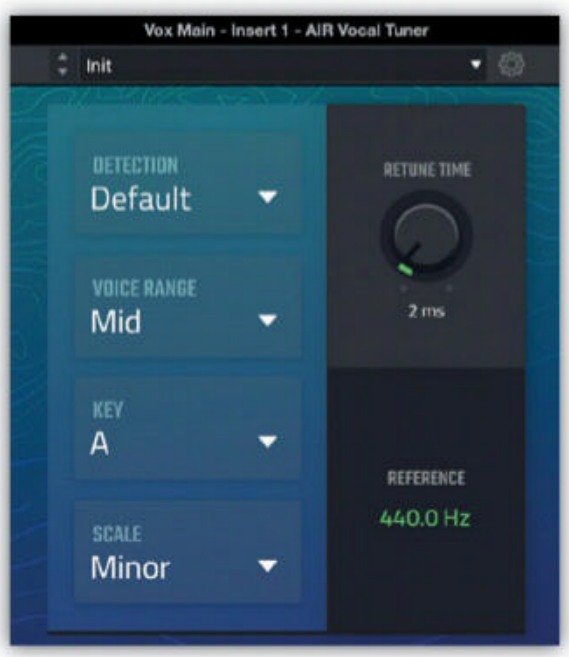
Force 10

Software development for the MPC platform has been gaining momentum in the last year or so, benefiting all users whether working standalone, hybrid or in the app or plug-in. We are now up to 2.10, edging close to the big 3.0, which I suspect will bring more parity with the Force, which was running a 3.x numbered OS from launch.

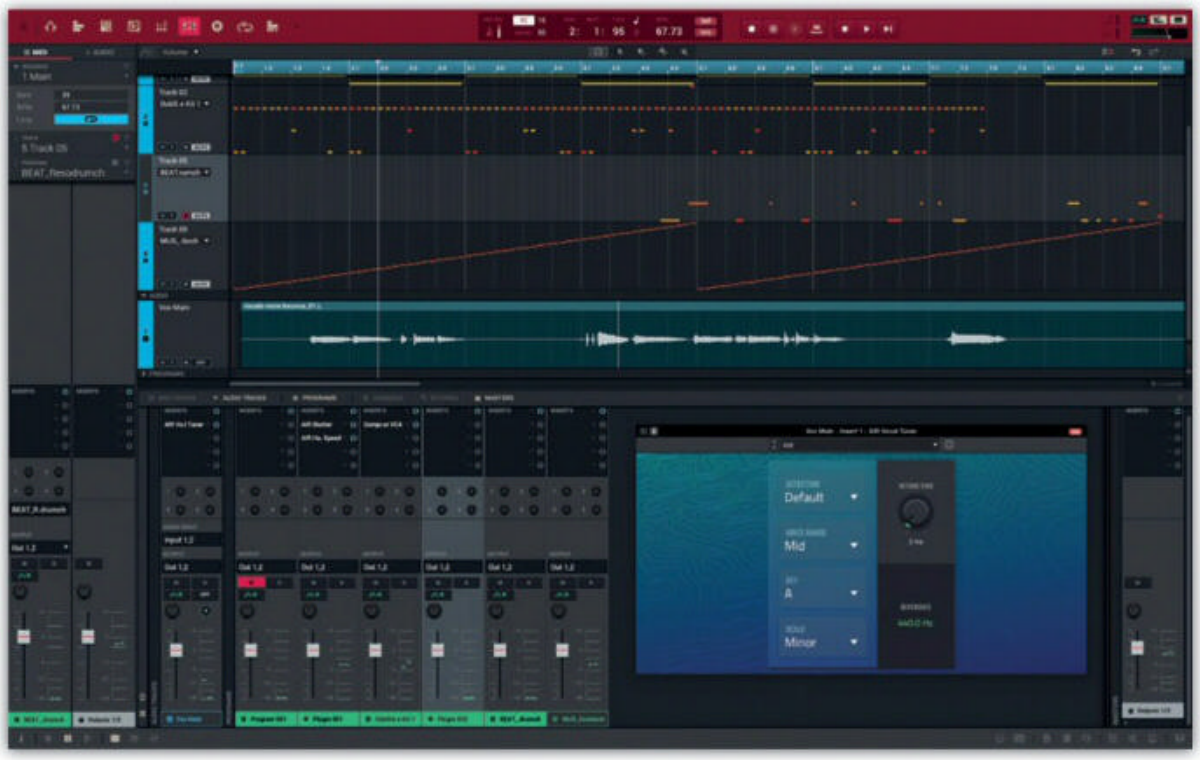
The standalone controllers got a great boost with the ability to connect USB audio interfaces, but the main meat of the update is a suite of new internal synths and effects. Adding to the drum synths that arrived in the previous release, there are four completely new instruments. Hype is the huge hybrid synth that powers Force, and has now finally come to the MPC. This is a super-versatile preset synth which combines virtual analogue, wavetable and sample sources with modulation and effects. Solina is a sample-based recreation of the classic string synth. Mellotron is what it says it is, combining the classic tape loop instrument with a variety of pedal-style effects. Finally, Odyssey is an emulation of the much-loved ARP synth.

There are also new audio effects, mostly from AIR, but also some Akai ones including a cool granular effect. Akai Pro know their market, and have hit on a winner with the new AIR vocal suite, which brings a doubler, harmoniser and a tuner to the MPC. Yes, the tuner will do the now obligatory hard auto-tune effect that at this point may never go out of fashion.

With the MPC Studio you are of course missing out on the hands-on experience of the custom touchscreen and Q-Link UIs that Akai Pro have created for all these instruments, but you do get to enjoy them as they are integrated into the MPC software. The MPC can host VST plug-ins too, in their own pop-out window, but the native devices have their own panels embedded in the lower pane of the MPC window.



Yes, the new vocal tuner plug-in does THAT effect.



The MPC software Tracks view, with one of the new Vocal effects.

a bit awkward: you need to go into a config mode, pick the encoder, then close an on-screen dialogue with the mouse. The touch strip works well for most parameters, but when I tried it for sample editing the tracking was too accurate and jittery.

What the strip is really great for are the Touch FX: the MPC Studio equivalent of the X-Y FX, reworked with new interfaces for single-axis touch control. Here you can play a suite of MPC performance effects, with the strip both activating and modulating the effect. There are versions of the classic X-Y effects like Beat Repeat, Beat Sync Filter and Tape Stop, but also some new additions such as Granulator and the excellent Washout.

Conclusion

Without the touchscreen or encoders of the larger, standalone MPCs, you'll need to be comfortable with using the software side to make the most of the MPC Studio. However, the Studio makes a lot of sense if you're already using the MPC/MPC Beats software, or if you mainly want to use MPC as a source of kits, instruments and samples in your DAW projects rather than as a dedicated workstation.

As well as providing a tactile focus of performance and navigation of the MPC software, you can toggle to a customisable general-purpose MIDI mode for playing other plug-ins in your DAW. It's a perfectly sized addition to a laptop setup, and the drum pads are the usual MPC quality.

£ £229 including VAT.
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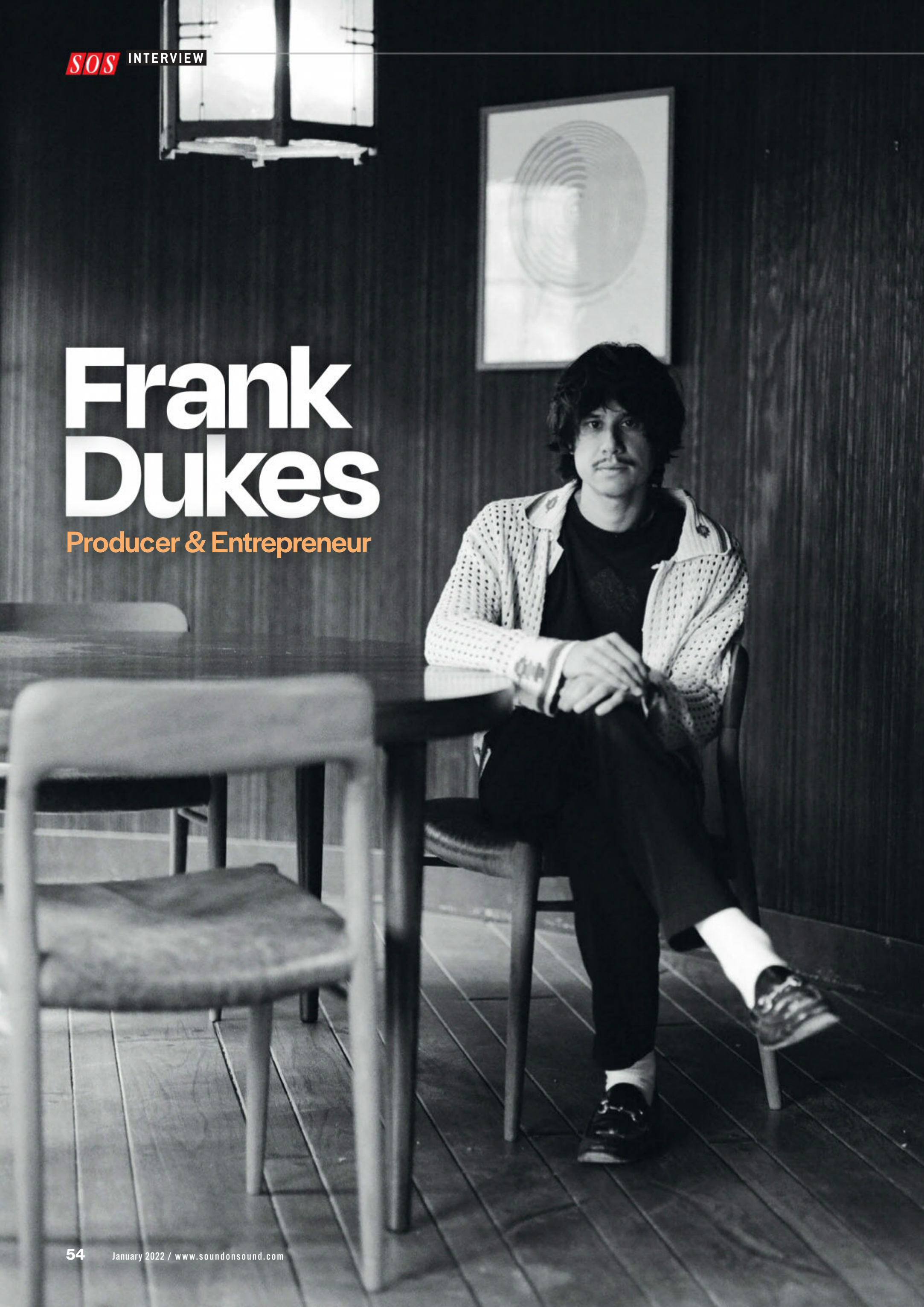
Frank Dukes

Producer & Entrepreneur



Having crafted some of the biggest hits in pop, Adam King Feeney, aka Frank Dukes, built an ambitious sample library to help others do the same.

PAUL TINGEN

"I love popular music, and I like music that veers on the stranger side of things. So I love pop music that veers on the stranger side of things! It's what my favourite artists like the Beatles, Nirvana and Wu-Tang Clan did. They made popular music that was atypical. As a result, I'm also known for making the stranger pop records. But they are not the consequence of a conscious decision to go out of the box. It's not intellectualised. It's simply me doing what feels exciting."

Speaking is Adam King Feeney, better known as Frank Dukes, the name under which he has operated since the early '00s. He has contributed to a swathe of major international hits by Drake, Eminem, Taylor Swift, Lorde, Kanye West, Camile Cabello, Post Malone, the Weeknd, the Jonas Brothers, and many, many more. Having cut his teeth in the hip-hop genre, Feeney went on to become one of the world's leading pop songwriters and producers. Since 2014, Feeney has won an astonishing 11 ASCAP Awards, 16 BMI Awards, three Grammy Awards, and 29 Grammy Award nominations.

Feeney says that his aim was "to be one of the best producers in the world and to work with the most exciting artists of our times," and the above suggests that his is very much a case of mission accomplished. He also clearly is proud of the fact that many of the records he's been involved in making have staying power.

"When I look at all my records, also my early hits, they were not meteoric, straight-to-number-one songs. Instead they started slowly, and made their way up to the top 10 gradually. But once they get there, they stay in the top 10 longer than many other songs. I'm happy to be the guy who makes slow-burning music that sticks around. I think there's something authentic in the records that I produce that maybe makes them resonate on a deeper frequency than some of the songs that were designed to succeed."

First & Last

Feeney has been invited countless times to share his secrets and process, but for some reason he has been reluctant. There are relatively few interviews with him online, and his appearance here is his first ever in a specialist music tech magazine.

Talking via Zoom from his home in Los Angeles, the Canadian traced his career so far. Feeney was born in Toronto, Canada, in 1983, received piano lessons when he was five, and went on to also play guitar and drums. In his teenage years he collected records, especially from the '60s and '70s, which got him interested in record production, and led him to buy an MPC. He also got into skateboarding and via that into hip-hop and DJ'ing.

In the early 2000s, Feeney took on the Frank Dukes name (based on a character called Frank Dux in the 1988 martial arts movie *Bloodsport*). He started working with rappers, doing remixes and getting placements, and he produced the track 'Money' for Drake. He also began a collaboration with producer Boi-1da, and ended up working with many different artists, most notably 50 Cent, several Wu-Tang members, and hip-hop/jazz group BADBADNOTGOOD.

Sampling Time

Frustrated by the practical and financial issues surrounding the clearing of samples, Feeney started to create his own, and in 2011 set up the Kingsway Music Library, which initially consisted of collections of his compositions that could be used as the foundation for samples. It has since become legendary, and has become a blueprint for how samples are made available and used, as well for the quality of samples in general. Countless artists and producers, amongst them Drake, Boi-1da, Vinylz, Kanye, Mac Miller and Rihanna have used Feeney's samples.

His next career move came in 2015, and followed a meeting with producer Louis Bell and Post Malone in 2015. It led Dukes to participate in the writing and producing of pop songs, with his name appearing on the credits of over 300 songs to date. Thirty of these have gone platinum.

"My career has taken many different forms over the years," Feeney says, "because I was just following what I was most excited by at any given moment. In the beginning it was just the very idea of being able to make music. Sitting in my parents' basement with my MPC making beats was amazing to me. Just the magic of starting with nothing and ending up with something. That was amazing.

"My next step was, 'OK, it would be nice to work with other artists.' So I get very excited about having rappers on my beats. Once I had done that, it was like, 'OK, what's next?' I had made a beat for 50 Cent using a sample by the Menahan Street Band from New York, and they invited me to New York. When I went over I saw this process that was for me entirely new: real musicians playing instruments and recording themselves to tape, using vintage equipment. The results sounded texturally just like the records that I had grown to love and that I had been sampling. That was eye-opening.

"When I get excited about something new, I just want to do it compulsively. So I got really into analogue gear and the process of recording on analogue. I maybe didn't realise this at the time, but I was beginning to understand how my favourite records from the '60s and '70s and '80s had been made, and the sounds I loved and had been sampling. Using analogue gear, I started making music that sounded similar, that I then used as samples.

"That was my obsession at the time. Eventually I was making more music to be used as samples than I was making beats, and I was sharing these samples with friends who were making beats. I set up the Kingsway Library to share these samples with everyone who was interested. This is how I got my first big hit record, when Boi-1da sampled a piece of music by me for the Drake song '0 to 100' [*2014*]. It led to a publishing deal and was the first time I made real money from music."

Feeney regards his contribution to the world of sampling as one of his greatest achievements. "Many producers said to me, 'You are known for your sounds, why give them to everybody?' But I love the idea of sharing. I just like to make things I am excited about, whether I make money from it or not. There's something unique to my experience that affects how I do things, and other people will use my sounds in different ways than I.

"I love the idea of making things that sound great and that ultimately contribute to the canon of music and makes music sound a little better. So I'm proud that what I have done with samples has contributed to evolving the musicality of hip-hop. Non-sample-based hip-hop music from before 2010 was much simpler and much less musical. The idea of making your »

Adam King Feeney's Lyric Studio in LA.

» own samples was new at the time. But today it's all over.

"There are these kids everywhere who are just loop or sample makers who send their stuff to producers. That did not exist. I see kids making great samples of things like full string sections and you almost cannot tell the difference between something sampled from old records and what's in the new music libraries. It's amazing.

"While the archetype of that kind music making was created by the concept I was exploring, I would never claim to be the first. The first person who I saw doing this was Nick Brongers, who was also referencing things that sound like old music. The innovation that I added was to focus on making things that sound new and that are ideas for beats, and not necessarily deal with old-sounding samples."

Influences

Older records nonetheless continue to be an important reference point for Feeney. When asked what records influenced him the most, he replies: "There are so many, I don't even know where to start. I love Arthur Verocai, the Brazilian composer who made a record in 1972 that was largely overlooked at the time, but when it was sampled by MF Doom [*in 2004*] it got really popular and has since often been sampled. To me the album is a masterpiece of songwriting, arrangement and production.

"My taste ranges from the most obscure to the most popular. The Beatles, Nirvana and Wu-Tang were all incredible and obscenely popular at the same time. My ears are definitely subconsciously attuned to good popular music, and it's why I write catchy songs. The Beatles evolved songwriting and the idea of what pop music can be, and they also evolved production techniques, doing incredible things with George Martin.

"The old music I love the most was pioneering at the time, but I would never recreate something to sound old just for the sake of it. I make music in 2021, not in 1960 or 1970. My perspective has always been to take things that I love and modernise them. That's what makes it compelling. No-one wants to hear what has been done."

Executive Decisions

In mentioning writing "catchy songs," Feeney has landed on what, until recently, was the latest phase of his career. "By the time I met Lou and Post, I had built up a catalogue of credits, and I had come to recognise certain producers as being really good at making drums and others at writing particular types of songs, and so on. This meant that I had been learning about true production and having an overview and bringing in certain people for specific things, while having control over what the final product would sound like.

"Producing and songwriting were whole new explorations that I got excited by. Up until that time I had not really co-written songs, but that transition was really easy, because I was already playing and writing music and making production decisions. Co-writing and co-producing songs for artists became my new obsession, and that transitioned into artists working with me for entire projects, and me executive producing their albums.

"My career evolved to a place where I felt I had arrived as a fully fledged record producer, and not just a guy who was only making beats or samples. For me that was a great arrival, because I felt confident to go into a room with any artist at any level, and be able to deliver. I have done so many different things and worked with so many different people and explored so many avenues, and they're all meaningful to me, because one cannot be without the other. I'm as proud of what I have done with the Kingsway Music Library, as of the records I have co-written and produced."

Waxing Lyrical

Until 2018, Dukes travelled regularly between his home town of Toronto and Los Angeles, where most of the records he was working on were made, but in that year he bowed to the inevitable and decamped to LA. In May of 2021, he moved again to his current location in the City of Angels, where he now lives on a property with several old cottages, one of them housing his Lyric Studio.

The studio gear at Lyric is as modern as one would expect, with ATC SCM45a monitors, UA Apollo x16 and Apollo x8p interfaces, tons of mic pres consisting a four-channel Neve Portico 5024, two Chandler TG 2-500s, two API 512c preamps, and two Neve 1073LB modules.

The vocal and guitar chain at the studio consists of a Townsend Sphere microphone going into an API 512c and then the Apollo x16. The piano is recorded with two AKG C414s going through the Portico 5024, and with regards to the drums, the kick is miked with an AKG D12, and the hi-hats with a Shure SM57, both going through the 5024. Snare top and bottom each have an SM57, through Chandler Germanium preamps, and overheads are recorded »

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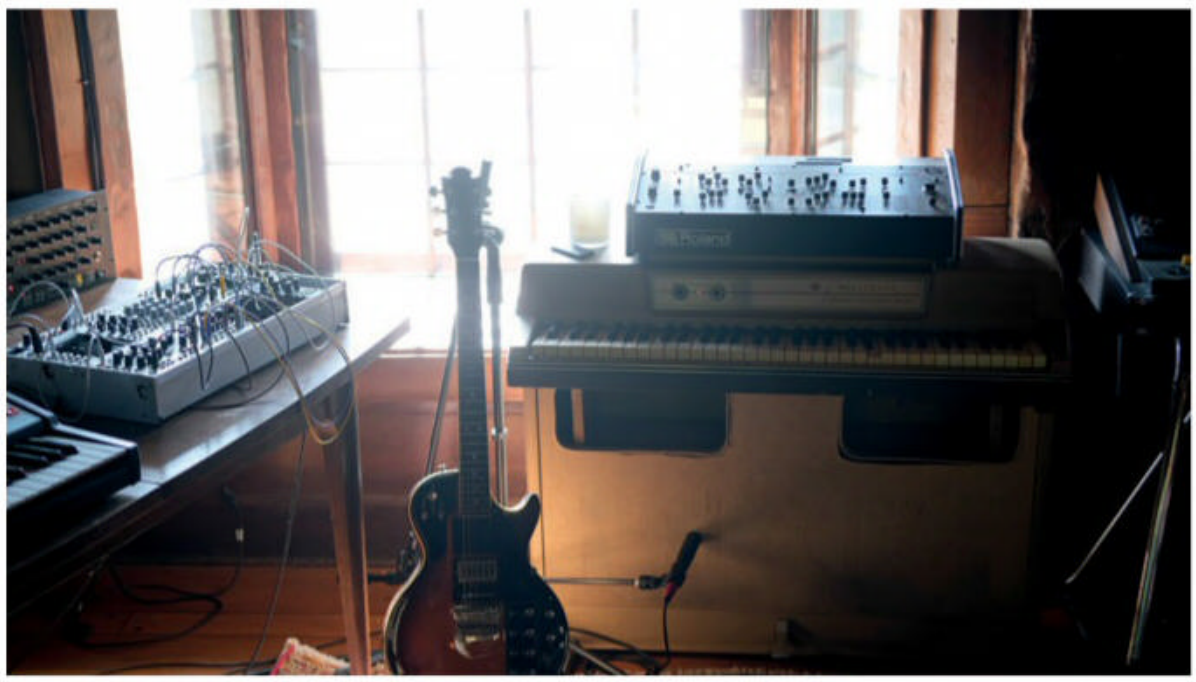
Feedey's analogue gear comes most into play with the extensive collection of keyboards and guitars at the studio. The collection includes a Yamaha CS60, Hammond organ with Leslie cabinet, Wurlitzer electric piano, Roland GR-500 Guitar Synth, Black Corporation Kijimi, Moog Matriarch, a modular rig, Sequential Prophet 10 and Prophet X, Korg Mono/Poly, UDO Super 6, Samick upright piano, and Elektron Digitone Keys, while the guitars are by Fender, Gibson, Martin and Hofner.

Feeney: "I have been collecting analogue keyboards and synths for over 15 years, and acquired tons of them. As I said earlier, excitement is the most important currency to me, and analogue keyboards definitely excited me! At one stage I became obsessed with analogue keyboards, and would be digging into Craigslist for deals, and looking at eBay, and going into local music stores, and I just bought anything that looked cool and made cool noises, and that I felt would inspire something.

"The first analogue keyboard that I bought was probably some crummy Crumar synth that has just three sounds, but I was blown away because it sounded like the records I loved. I think the next synth was a Roland Juno-60, and then a Yamaha CS101. The first really big synth I got was a Yamaha CS60. But I am less precious about them at this point. When I was younger, it all had to be analogue, but today it does not need to be so extreme."



Over the years, Feeney has amassed an enviable collection of analogue keyboards.

From A To D

In fact, Feeney appears to have completely released his attachment to analogue gear. "I am no longer an analogue purist. As I have grown and become wiser, I have realised that analogue is not necessary. It is not about analogue or digital, it is about inspiration. It is about what makes you excited to make music. I still love analogue recordings, and I still love the process. I love what it does psychologically, for example the fact that you don't have to sit behind a computer, which feels more inspiring to me. I definitely continue to gravitate towards things that sound more analogue, but that does not necessarily mean that it has to be done in analogue.

"There was a time when I was like: 'If I'm working with a band, it has to be done on analogue.' But today I cannot imagine recording to tape anymore. I feel that digital technology and my skills have developed to the point where I don't think anyone could tell the difference. You can work in digital and still capture that analogue spirit. There are plug-ins now that truly make things feel and sound analogue, like the Waves J37 Tape Saturation, which is probably my favourite plug-in.

"Also, the analogue synths that are coming out now are so much better than the old ones. I get that you want to buy vintage equipment from a collector's point of view. But as a musician I'd rather buy the new version of a synthesizer, that sounds

incredible, has MIDI, and you know is not going to break, and that it has all the modern functionalities to incorporate into a modern setup. New analogue is in an awesome place now."

DAW Of A New Era

At the heart of Feeney's digital setup is Ableton Live. "When I started I just worked on MPC and vinyl records," he remembers. "In 2007, I discovered Ableton, and I initially used that to record beats from the MPC. Then Ableton became the 'brain' of my setup as opposed to the MPC, and I have used Ableton ever since.

"There were things that Ableton had not worked out yet in the past, but its functionalities are getting much closer now to those of other DAWs. In any case, I'm pretty primary in terms of exploring all the functionalities of the software. I use Ableton mostly like an extremely advanced tape machine. It's the intention and what you're doing that matters, not the software.

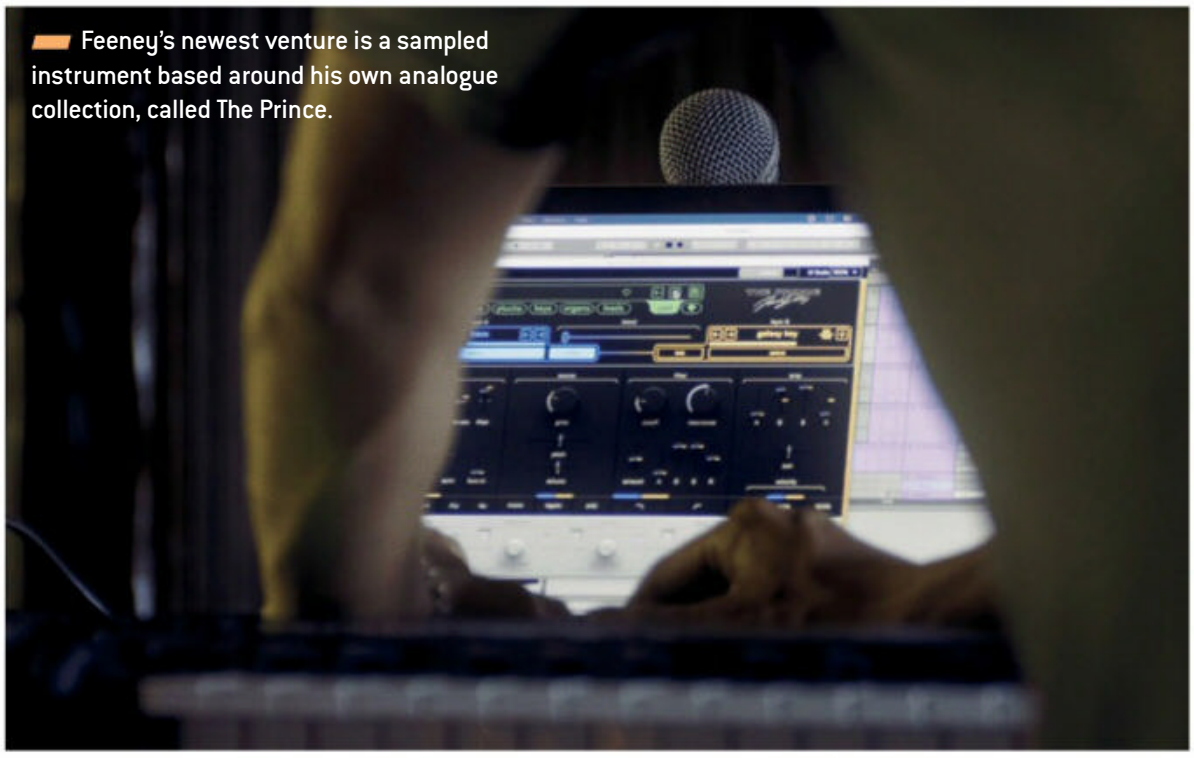
"When I make music my process is very simple and intuitive, and I just follow what excites me, and go with the flow. I can start on the piano, but when I write a song I'm usually on guitar. My engineer, Tyler Murphy, records that stuff into Ableton, and then he places it onto a grid, and we go into time-stretching and things like that. We love time-stretching in Ableton and how natural and intuitive it is. I may then add a beat or play drums. We have developed a really great process for stretching them by using Ableton's ingenuity.

"I'm a firm believer in the idea that as a record producer, you do not need any technical knowledge. You just need to be good at making decisions. Tyler is like the technical wing of my brain and an extension of me. When he's around, I don't have to think about the technical side of things, and can just think about being creative."

The Prince

Ever searching for new things to get excited about, Feeney landed on The Prince. The idea for his brand-new sample-based VST synth was initially born from a need that arose when he was still regularly travelling between Toronto and Los Angeles, and it was complicated to take his analogue keyboards, amps and pedals with him.

"I was looking for a way to bring my own sounds with me in one small, portable box. So I started this exploration, working with my friend Matt Fudge who is a software programmer and developer. We realised that there are very few synths with a sample-based architecture. We could either use Native Instruments' Kontakt, or build something from the ground up. We chose to do the latter, so I set up a company, called Cradle Apps, and put together a team to build something substantial that could truly evolve the state of sample-based virtual instruments.



Feeney's newest venture is a sampled instrument based around his own analogue collection, called The Prince.

"We started on this three years ago, and it's been an ongoing project on which we chiselled away until the release last October. It took long to get it right! We ended up with something that I'm really excited about. The Prince is based on my 20 years of collecting sounds. It takes sounds directly from all my favourite synths, and all the sounds I have used on famous records I have been involved in. They're all in there. I have curated every sound.

"The Prince is a documentation of my sound selection throughout my career until now. We really homed in on what makes analogue synths sound great, and I think the engine we developed sounds like no other sample engine. We figured out how to capture that baseline noise that analogue synths have, making sure we don't amplify it, but also not getting rid of it, as most other sample engines do. You can use The Prince as a preset machine, and never get tired of it. But you can also do your own sound design and make insane sounds.

"The design is very simple and very intuitive. I wanted to create a synth for which you do not need any technical knowledge. The ethos of the entire company is to make products that sound incredible and inspiring and for which you do not need a lot of technical knowledge. You load up The Prince and it sounds incredible, and it makes you want to play things and write music. That initial feeling it gives you the second you open it up was really important to me.

"Honestly, The Prince is now my favourite VST, and it has really changed my process. I essentially put all my favourite sounds in one place, and together with the [*Waves*] J37 it's all I use. It's almost comical. I get questions all the time about what I am using to get my sounds, and then they see me in a session, and notice it's just The Prince and the J37, and they go: 'That's it?' As I said, I'm a simple guy and gear is not that precious to me."

What's In A Name?

Finally, about that career change. At the time this interview went to print it was still under wraps, and all Feeney could say was that he'll continue to produce, but that he'll retire the Frank Dukes name because in 2022 he'll be shifting his focus to a new arena.

"About three years ago I felt that I had achieved everything I wanted in terms of my dream to become a top producer, and I started wondering, 'What's the next thing for me to get excited about?' As a producer you are of service to the artist. You serve their vision. I can inject pieces of myself into that, but ultimately, what makes a good producer is their ability to serve the artist's vision.

"I'm now at a point in my life where I want to realise my own vision. The next phase of my career will be about sharing my story, and about giving people things, like The Prince. My career as a producer has often been about surprising people, and I hope my next steps will be as well."

Zeppelin Design Labs Cortado MkIII

Contact Microphone

What would you record with an indestructible microphone?

MIKE SENIOR

For the most part, microphones are designed to pick up vibrations in the air that have been created by some kind of physically vibrating sound source. But there's one specialist mic design, called a contact mic, that can cut out the middleman (ie. the air) by directly transducing the sound source's physical vibrations into a voltage signal.

To be frank, though, contact mics have always seemed pretty niche to me as far as everyday recording tasks are concerned. I remember buying one out of sheer curiosity about 15 years ago, but it was a fiddly, cheap-looking thing that was noisy as hell and made my violin sound like a cat in a biscuit tin. As such, I'd mentally filed contact mics under 'break glass in case of drum triggering' ever since.

A couple of months ago, however, my interest was rekindled by a new product from Chicago-based company Zeppelin Design Labs: their Cortado MkIII, a contact mic that aims to elevate this transducer concept into a bona fide recording tool. But does a contact mic really deserve a place alongside traditional mics in the studio? Let's find out...

Hardware

The business end of the Cortado MkIII looks, for all the world, like one of those little tins of lip-salve, measuring about 36mm in diameter and 14mm thick. A slim 185cm shielded cable sprouting from the tin's side connects its internal vibration transducer to a Class-A preamp, housed in a separate 43 x 43 x 83 mm steel box. The preamp's active electronics receive phantom power from the box's single male XLR socket, which also serves as the balanced audio output.

On the underside of the preamp are four miniature DIP slide-switches, two of which can activate a -10dB input pad, while the others determine whether the preamp's internal high-pass filter rolls off at either 23Hz or 150Hz. The Cortado's published frequency response is an impressive 23Hz-40kHz (at the -3dB points), and the provided plots show that the sub-100Hz region gets a couple of decibels warmer when the pad's engaged.

The whole package feels neatly put together, and the mostly metal construction seems pretty robust too. Indeed, the manufacturer's product video shows the transducer being run over by a car without damage, and I can well believe it. Furthermore, the mic's operating temperature range is listed as -40 to +80 degrees celsius, and you can even use the transducer underwater, so it's clearly no shrinking violet! I do question the use of DIP switches for settings that you're likely to be changing day to day, though. DIP switches seem better suited for set-and-forget parameters, and I'd rather Zeppelin had added another $10 to the asking price and provided more substantial switches that I could operate without taking the toothpick out of the corner of my mouth...

There are lots of ways you can fix the Cortado to a source of vibration. For a temporary solution, the mic ships with a chickpea-sized blob of sensor putty that can stick the mic to most surfaces. (Despite its technical-sounding name, my 'sensor putty' was supplied on Faber-Castell backing paper, which suggests that further supplies may not be too hard to come by!) For a more secure solution, Zeppelin suggest those little plastic spring clamps you use for DIY, but while those work well where you've got something fairly thin and flat to clip the mic to, I found that mini bar-clamps provided more scope for experimentation, given their much wider jaw span. For permanent mounting, Zeppelin offer an optional metal screw-on bracket, but also recommend strong double-sided sticky tape as a less heavy-duty alternative.

Contact Sport

If you read *SOS* regularly, then you'll likely already know a thing or two about mic technique. But I suspect it won't help you much here, because a contact mic is actually a very different animal!

Zeppelin Design Labs
Cortado MkIII

£119

PROS

- Overflowing with cool indie/retro sounds, both for traditional instrument recording and creative sound design.
- Can turn pretty much any object into a microphone.
- Solidly built with balanced output and decent tech specs.
- Spill-free recording of acoustic instruments for ensemble-recording scenarios or sample triggering.

CONS

- Mic positioning involves more trial and error than with traditional mics, and specific sounds are less repeatable between sessions.
- As with traditional close-miking, general-purpose sounds can be hard to achieve without multimiking.

SUMMARY

The Cortado MkIII offers a whole world of cool-sounding retro/indie timbres to the creative recording engineer. It can take a little work to get the best out of it (and it works best when multimiking), but it's bags of fun and excellent value for money.

Although Zeppelin suggest using a uniform layer of sensor putty to fix the Cortado to an instrument, I found that a 'bullseye' pattern actually delivered more dependable results.

For a start, the mechanical operation of some musical instruments can create physical vibrations within the instrument's body that wouldn't normally impinge heavily upon the listener (or a normal microphone), but which dominate over the instrument's tone if you're contact-miking. So you can pretty much rule out getting any kind of natural balance on woodwind or brass instruments with the Cortado on account of key/valve clunks, for instance. My upright piano's pedal action also came through more prominently than I'd have expected with traditional mics.

There is one sense, though, in which using the Cortado could be seen as an extension of close-miking. You see, when you put an SM57 an inch away from your snare drum, you're mostly going to pick up just that small subset of the instrument's frequency output that emanates from the coaster-sized surface area the mic's actually pointing at. This is why moving the mic just a little bit under those circumstances can make such a big difference to the sound. With the Cortado, though, the transducer's contact plate picks up vibrations from an even smaller surface area (and more from the centre of that area than from the edges, it seems to me), so the changes in tone you get by moving this mic just an inch can be crazy.

Furthermore, the exact way the transducer is coupled with the vibrating surface can make a huge difference to the resultant output level and tone. This makes using the sensor putty (as I suspect most users will) a little bit of a lottery. For instance, when I repeatedly remounted the mic in an identical location (as close as I could judge it) alongside the bridge of my acoustic guitar, I kept getting audible timbral variations — the amount of low end in particular seemed to be quite fickle in this respect. (The Cortado's user manual recommended using a uniform layer of putty across most of the contact surface, but for my part I found that a kind of 'bullseye' pattern seemed to give a slightly more predictable tone.) Using clamps instead of putty made things a lot more repeatable, especially when using the bar clamps, since these were able to press the transducer down more firmly.

"The Cortado is an absolute gem... it offers a wellspring of exotic, characterful, and unrepeatable sounds that go well beyond what traditional mics can offer."

The upshot of all this is that finding a suitable sound with the Cortado usually takes a good deal more trial and error than when you're using normal mics. And even when you find a mic position you like the sound of, if you're working with the sensor putty then there's no guarantee you'll be able to recreate that sound again on a future session — or indeed if the player accidentally knocks the transducer between takes! To be fair to Zeppelin, though, I wouldn't say these are criticisms of the Cortado per se, because such difficulties are inherent with any contact mic, as I see it. And there's a silver lining too, in that you can also deliberately tamper with the transducer coupling for creative tonal ends, perhaps by sandwiching a layer »

The Cortado can pick up vibrations from almost anything, so it's great for creative sound design.

» of some different material between the source of vibration and the mic.

Another practical consideration is what to do with the tether cable from the preamp, because it will happily carry vibrations to the transducer too, and I did on occasion get undesirable buzzes coming through on the signal simply because the cable had come into contact with some vibrating component of the instrument. In addition, if the cable is pulled taut, it may interfere with the transducer's own vibrational characteristics, effectively applying a kind of mechanical filter, as well as potentially altering the transducer's surface coupling. So having a couple of spare clips and a rag to damp the cable and hold it out of harm's way isn't a bad precaution.

Listen Online

There's only so much you can do to characterise a mic's sound in print, so I've set up a special web page to showcase Cortado recordings of various different instruments and improvised resonators. I've also done a selection of audio examples showing some of the effects of positioning and transducer-coupling changes, as discussed in the review. Oh, and you can even catch my incomparable stereo toast Foley...

W www.cambridge-mt.com/misc/zeppelin-cortado/

Studio Tests

As you'd expect, the Cortado's basic sound on instruments like acoustic guitar and bass has the same kind of super-dry compactness you'd associate with a regular pickup DI signal. But to my ears the mic also captures a lot more of the unique resonant character of the instrument by comparison, so its timbre always feels somehow richer and more full of personality, if inevitably less predictable in terms of frequency response. Furthermore, moving the Cortado's contact position and coupling method delivers a vast menu of evocative tone colours. This isn't a mic I'd choose for uncomplicated transparency, but if you want a sound you'll really care about, something with bucketloads of earthy musicality, then the Cortado delivers that in spades! It just feels like everything that comes out of it belongs on some kind of fantasy Tom Waits record co-produced by Mitchell Froom, Tchad Blake, and Joe Henry for the Nonesuch label. (If none of those names mean anything to you, then this might not be the kind of product you're looking for...)

Furthermore, the strongly coloured tonalities this mic typically produces can frequently be made a lot more general-purpose by combining signals from multiple Cortados, in much the same way as you might construct a more representative close-miked snare sound by multimiking the drum from different angles. In fact, this improvement felt so striking when contact miking, that I reckon you'll get significantly better value for money from a pair of Cortados than from a single one — even setting aside the issue of stereo capture. (Which I really shouldn't, given some of the genuinely wonderful bone-dry panoramas a pair of these mics have conjured up for me during the review period.)

All this good stuff is, however, only one facet of the Cortado. For a start, its ability to extract sound from all sorts of unlikely physical objects will doubtless appeal to anyone working in sound design. For instance, I recently managed to contact-mike a bit of toast while it was being spread with butter! But this same attitude also feeds back into the music-recording world once you start harnessing the idea of sympathetic vibration. A pair of Cortados on a window, floor, or radiator will catch a drum room sound that's unlike anything normal mics would, because of the physical filtering and resonant properties of the different materials involved. Again, it most definitely won't sound transparent, but who cares when it sounds extremely cool?

And you don't have to extend this concept too far to realise that the Cortado lets you effectively make a quirky-sounding 'traditional' mic out of practically anything. Zeppelin have videos showcasing Cortado-based vocal mics they've made out of a tin can and a plastic dish, and I had a blast recording

Little woodworking spring clamps (left) are good for clipping the Cortado to thin objects, but the wider jaws of lightweight bar clamps (right) were more useful for thicker objects such as the body of an acoustic guitar.

vocals with a lampshade, a glazed cabinet door, a stainless-steel cat dish, a plastic chopping board, an offcut of vinyl flooring, and an upright piano with the pedal held down! Now, admittedly, background hiss can become an issue in these applications if you're trying to record quiet sounds through recalcitrant materials, but to be honest that noise usually felt totally in keeping with the often retro-tinged sonic outcomes, so I don't imagine it'll much bother anyone who fancies diving head-first into this kind of stuff. And I should add that when miking any instrument directly, noise was never an issue for me — the Cortado is fundamentally a high-output, low-noise design, and I actually think it's quite impressive how good a job it does of making low-level sympathetic vibrations accessible and usable for music-recording purposes.

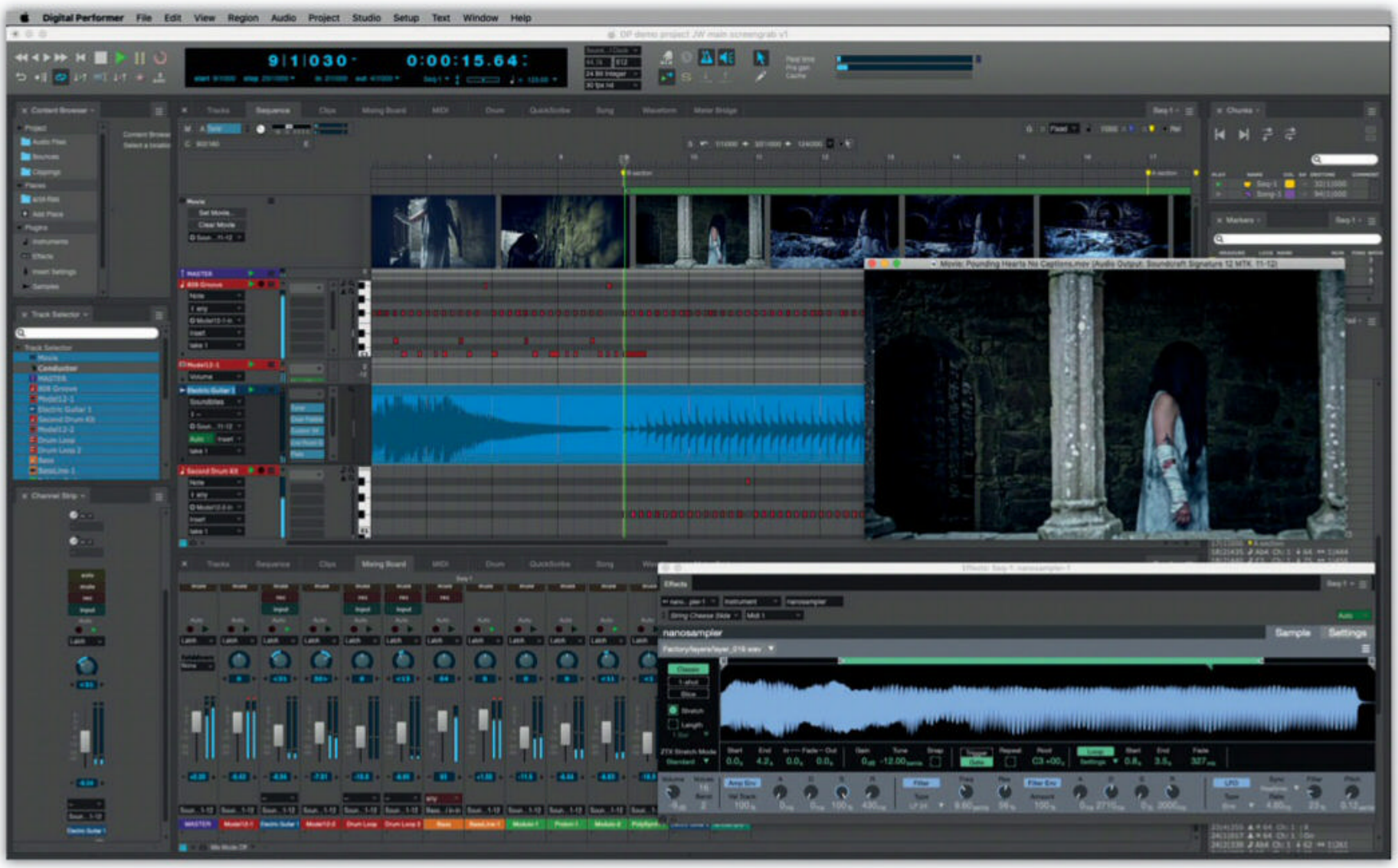
First Contact

I certainly wouldn't recommend the Cortado MkIII as the first mic you buy, and I also imagine it's unlikely to appeal to engineers who predominately aspire to the faithful capture of acoustic reality. But if you already have a few mics that cover most run-of-the-mill studio tasks, and want to massively expand the tonal palette at your disposal, then the Cortado is an absolute gem. For a little more than the price of a Shure SM58 (and considerably less than the price of the frequently maligned AKG C1000), it offers a wellspring of exotic, characterful, and unrepeatable sounds that go well beyond what traditional mics can offer. To my mind, it's already worth the price of admission just for what it can deliver when contact-miking instruments directly, but on top of that there's the whole world of sympathetic resonance and DIY mic design to explore, not to mention the simple utilities of spill-free capture and trigger-signal generation.

Clearly, some of the Cortado's capabilities are common to contact mics in general, and you'll find no shortage of much cheaper (and indeed physically more slimline) alternatives online. But I can't say any of those tempt me very much, when compared against the Cortado's solid build, balanced output, and commendable technical specs. That said, if you really can't conscience paying this much for a contact mic, then you might want to check out Zeppelin's own self-build kit of their previous-generation Cortado MkII, which retails for about a quarter of the price of the off-the-shelf MkIII. Knowing the extent of my own soldering proficiency, however, the price-differential in my case probably wouldn't cover the inevitable medical bills, let alone the additional costs of on-going trauma therapy...

Overall, then, I think the Cortado MkIII is a fantastic little product, and well-priced for what it offers. In case it's not clear from what I've written so far, I've not had as much fun with a new mic in years — and it should probably come as no surprise that I've already bought a couple of these beauties for myself!

£ £119 including VAT.
T breakthemachine.co.uk
W www.zeppelindesignlabs.com



MOTU Digital Performer 11

Digital Audio Workstation

DP11 might not be any louder than DP10, but it's got plenty more to offer.

JOHN WALDEN

MOTU's Digital Performer is a well-established product within the DAW/sequencer marketplace. It initially found favour amongst media composers with ahead-of-the-game support for working with video. However, over its 30+ year lifespan, it has gradually evolved into a deep, powerful, cross-platform (Windows support appeared alongside the Mac OS version in 2016) music production environment with a comprehensive feature set to appeal to almost any type of music maker.

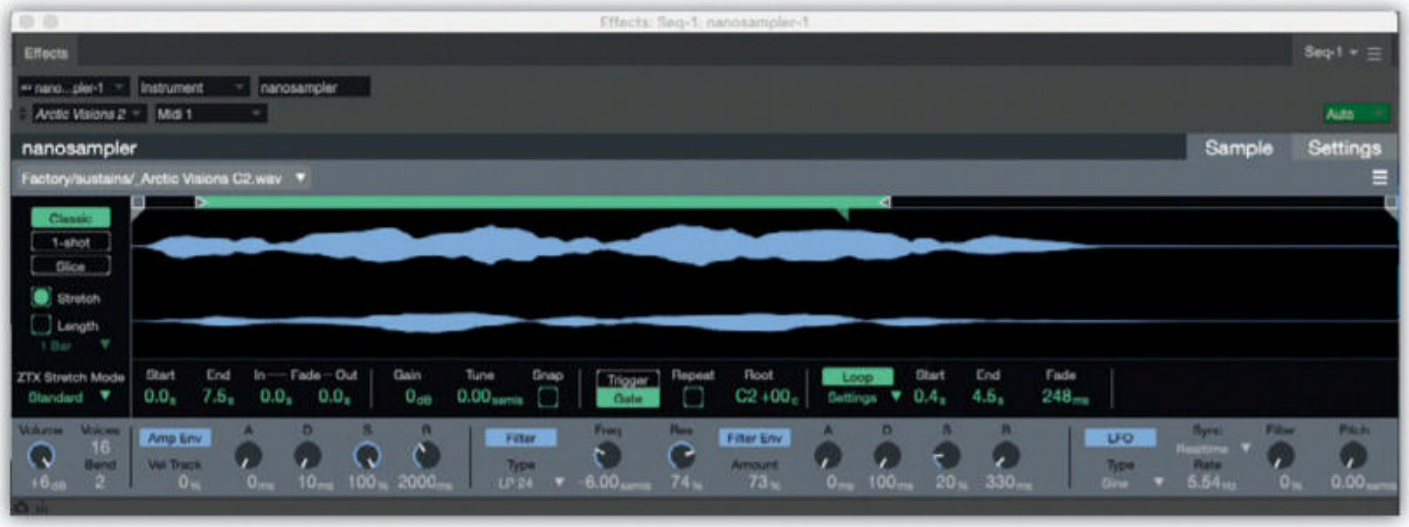
Digital Performer 11 is now with us (indeed, the subject of this review is actually v11.02) so what have MOTU added to turn up DP's 'buy me!' factor to one louder than 10? Let's find out...

Count To 10

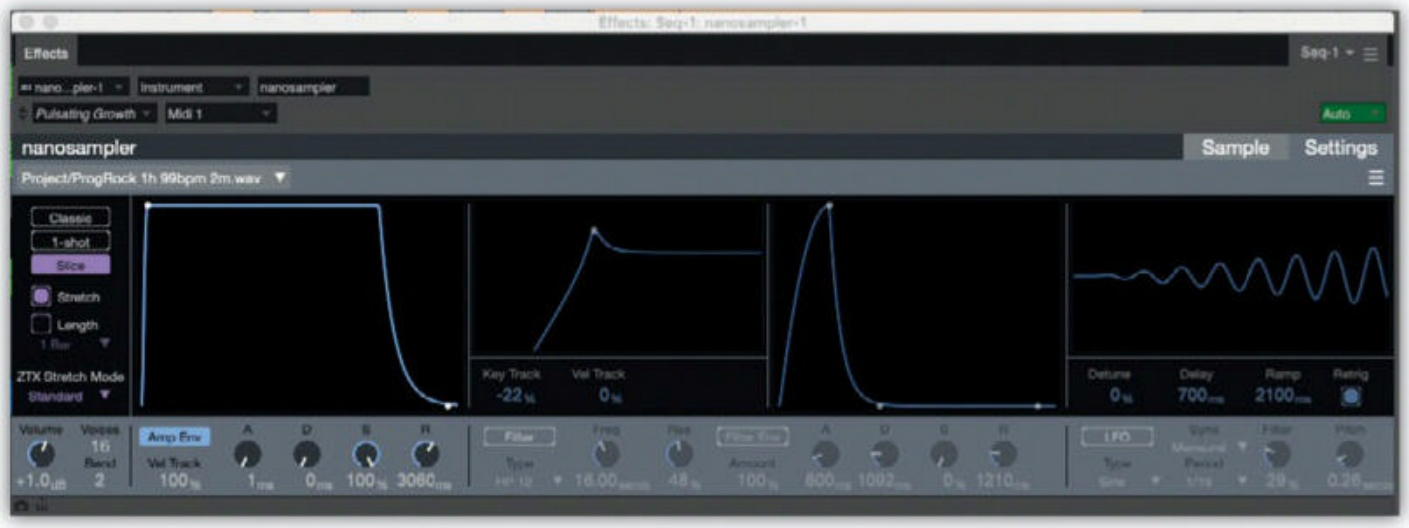
SOS reviewed DP 10.1 in the April 2020 issue. As briefly summarised there, DP's feature set most certainly makes it a worthy member of the DAW/sequencer elite. And, while it might have built its initial core following amongst film and TV composers, features such as the Clips Window — added in v.10 and very much aimed at using DP in a live performance context — clearly demonstrate MOTU's desire to keep expanding DP's appeal. DP is undoubtedly a music production environment with very many strengths.

However, there is no single DAW/sequencer that's perfect for every user and, as with any of DP's obvious competitors, it's possible to point to some areas that might benefit from some extra development. With DP, perhaps the most obvious gap has been the bundled virtual instruments. Prior to v.10, options such as BassLine, PolySynth, NanoSampler, Modulo, Model 12, MX4 and Proton were usable if perhaps not pushing any boundaries. DP 10 brought two further useful additions on this front. First, enhanced compatibility for third-party virtual instruments was provided via VST3 support. Second, a special 5GB 'MOTU soundbank' was added courtesy of a collaboration with UVI (and that utilised the free UVI Workstation playback engine). These were both positive steps but perhaps also still left scope for further improvements.

So where does DP11 take us? Given the mammoth feature set (DP's PDF manual runs to over 1000 pages), for the sake of brevity (and trees), we will focus here on what MOTU have added or improved for this latest release. Those unfamiliar with DP can, of course, delve into the excellent *SOS* archives if they wish to dig a little



Nanosampler's UI has been completely revamped for its 2.0 iteration.



Nanosampler's Settings panel provides graphical editing for the various modulation envelopes.

deeper into the history and evolution of DP's (comprehensive) specification.

Nano Newness

Given the observations above, perhaps one of the highlights of the DP11 release is a significant revamp of the Nanosampler virtual instrument. While you are still limited to working with a single sample in any instance of Nanosampler, you now have a very useful extended array of features for controlling how that sample is manipulated and triggered for playback.

All this extra functionality has also meant a comprehensive overhaul of the instrument's UI. This features browsers for both factory presets and samples (Nanosampler ships with a good crop of both) at the top of the display, a much more sophisticated waveform display in the central strip with the Global Settings panel on the left, and a bottom-most strip that contains a more expansive set of amp, filter and LFO controls. Two buttons located top-right can be used to toggle the central part of the display between the waveform view (Sample) and an envelope view (Settings). The latter gives you graphical control over the amp, filter and LFO properties.

Nanosampler 2.0 offers three main playback modes; Classic, 1-Shot and Slice. Classic maps your sample across the MIDI keyboard and can be used with or without sample stretching. There are various stretching modes with and without formant correction so you can experiment to find which produces the best results for any given sample. Classic mode also allows you to define a loop region within the sample for creating sustained sounds. The various envelope options and the LFO allow you to add modulation to your sound without getting bogged down in parameter overload. The filter itself offers various low-pass, band-pass and high-pass options. These are all very effective, but it would be even better if the filter perhaps offered a few more creative possibilities such as a distortion mode or two with modulation. That said, if you want to spice up your Nanosampler creations further, DP11's effects plug-in suite provides plenty of further processing possibilities.

As the name suggests, 1-Shot simply triggers the sound once and is suited to things like sound effects or individual drum hits, while Slice mode does pretty much what you would expect to a drum (or other rhythmic sound) loop and maps the individual slices across the MIDI keyboard. You get various options in terms of the slicing process and sensitivity, as well as options for editing individual slices. It all works very well and provides an effective way of getting more mileage out of your drum loop collection. The Randomise option for slice triggering is worth a particular mention. It can randomise the slice mapping to produce some cool rhythmic variations in real time on playback and can be applied to just selected slices if you want to introduce just a subtle amount of variation. My only (minor) complaint is that there doesn't seem to be a way to highlight a slice within the waveform display as it is triggered. This would make it much easier to navigate your slices during both playback and editing.

Other than that, in use, Nanosampler 2.0 can be a lot of fun and it is possible to create some very playable sounds. While it is not about to become a competitor for a dedicated third-party, multisample, multi-velocity-layer, sampler instrument, in terms of creating perfectly playable instruments out of single samples, or slicing a drum loop for reworking, it strikes an excellent balance between features (enough to provide creative »

MOTU Digital Performer 11
£397

PROS
- Some very useful improvements for media composers.
- Always-on audio recording can be a lifesaver.
- Evolution rather than revolution (and that's a good thing).

CONS
- Nanosampler 2.0 is great, but you will still want to supplement DP's virtual instruments with some third-party options.
- Deep and powerful, but with a significant learning curve to new users.

SUMMARY
This is an evolutionary rather than revolutionary upgrade, but MOTU have continued to refine the DP feature set is some genuinely useful areas that should particularly appeal to media composers.

Slice mode lets you get creative with your drum loop collection.

flexibility) and ease of use. As with the Sampler Track in Cubase (and to which Nanosampler 2.0 is conceptually very similar), don't underestimate what you can do from a single sample; Nanosampler 2.0 has plenty to offer for DIY sound creation.

Always-on Audio

Like most major DAW/sequencers, DP has offered retrospective MIDI recording — allowing you to capture that last bit of brilliant MIDI noodling even when you were not actively recording — for some time. By default, this process is always active in DP for any MIDI tracks that are record-enabled, whether the project is in playback or not. New in DP11 is an equivalent feature for retrospective audio recording. Again, this works for any record-enabled track and, whether the transport is running or not, DP11 is recording audio signals in the background. If you suddenly find yourself wishing that you had actually captured what you were playing or singing, then you can now retrieve it.

Whether MIDI or audio, the Retrospective Record sub-menu is available from the main Regions menu. What happens when you execute the Retrospective Record Audio command depends upon whether playback was operational or not and various options are available in terms of the placement of the retrieved soundbite on the timeline. By default, the most recent 'non-silent' section of your audio performance is what gets placed (that is, from the last pause in the audio signal). However, rather wonderfully, you can edit the left edge of the soundbite in the normal fashion to reveal earlier parts of your performance. How far back your audio time travel can go depends upon settings within the Play And Record section of the Preferences And Setting dialogue, but the defaults on my system were 5 minutes and a total of 1GB of memory was reserved for that purpose. You can, of course, configure these settings to suit your own needs or disable retrospective audio recording entirely if system resources are limited.

In use, I found this feature to be very impressive. There are always occasions when you are rehearsing a part prior to recording and nail it, only to then waste lots of time fluffing it up when recording is actually in process. This new feature can pretty much eliminate that frustration providing you have the required track record enabled.

Artful Articulations

While articulation keyswitching is something that many third-party sample libraries provide, media composers will undoubtedly welcome the addition of Articulation Maps within DP11. This allows you to define your own combination of MIDI notes for switching between performance articulations within a multi-articulation sound source. This is, of course, going to have particular potential for those working with string, brass or woodwind sounds in any of the more comprehensive orchestral libraries.

Such articulation switching can, of course, help make your orchestral (or other instrumental) mockups more dynamic and realistic. However, for those then generating printed musical scores to be used by real players, the additional benefit is that the articulation changes created by an Articulation Map can also generate suitable articulation symbols within the QuickScribe Editor. Articulation editing can be done within the QuickScribe Editor or within track lanes in the Graphic Editor.

The new Articulation Map feature has plenty of scope for customisation.

Clearly, to realise this potential, you need a suitably configured Articulation Map for the multi-articulation instrument patch in question. In use, creating an Articulation Map for a simple ensemble strings preset, and featuring sustained, staccato, pizzicato and tremolo articulations, took me just a few minutes. Having then selected the newly created Articulation Map within my MIDI track, I was then able to display articulation lanes for the track within the Sequence panel and assign/edit articulation changes as required. Flipping to the QuickScribe display showed the articulation symbols were then present within the notation. For the occasional articulation change this works well enough but, for more complex/rapid changes, there would be some careful formatting required before you might feel confident putting the score in front of a session musician (or 50).

While it requires a little effort (and perhaps a little head scratching), the

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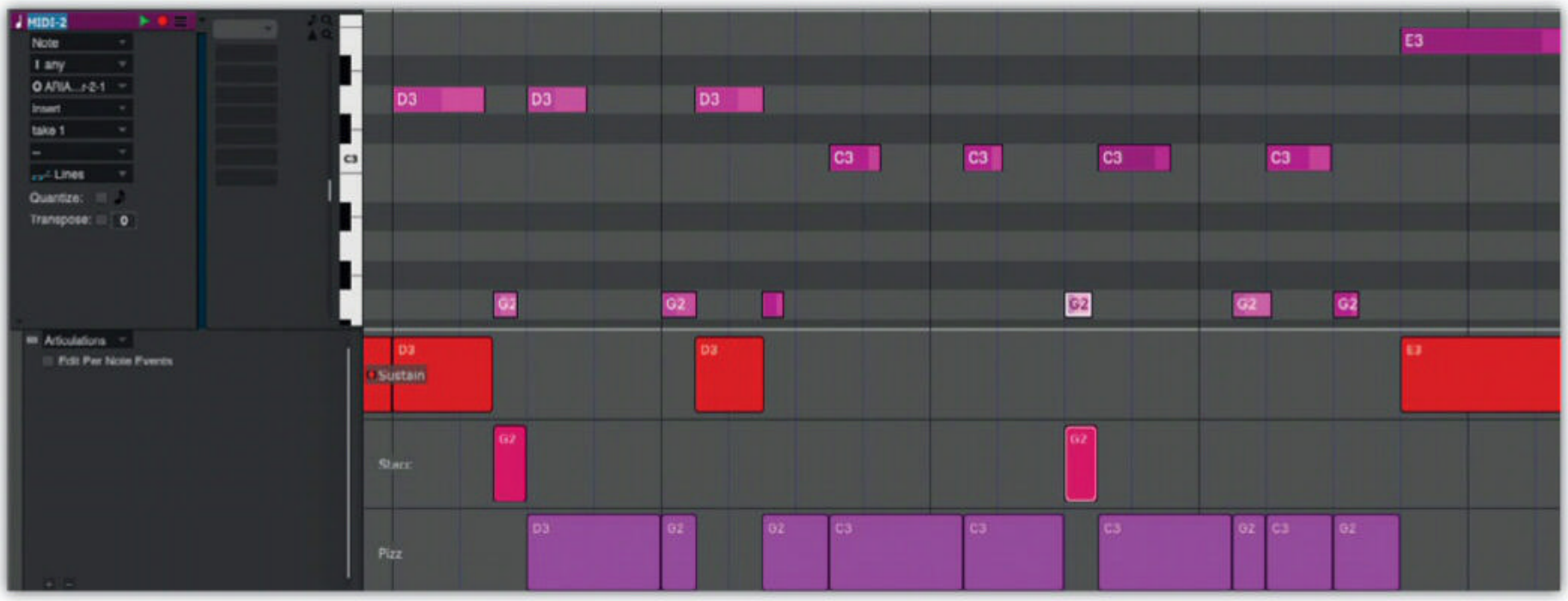


Once a MIDI track has an Articulation Map assigned, you can edit articulation changes within a dedicated lane.

» Articulation Map Setup dialogue has some interesting options. As well assigning the appropriate MIDI note (keyswitch) to each articulation, the Output Settings section lets you also add automatic adjustments to MIDI note data for specific articulations. For example, you can apply scaling to MIDI velocity or note length. You can also assign patch changes or MIDI channels to specific articulations so, for example, if you prefer to work with single articulation presets via a multitimbral virtual instrument (with different performance patches on different MIDI channels), you could then switch between these sounds in real time from a single MIDI track. Usefully, articulation switching also offers three modes — Permanent, Temporary (making it easy to switch back to a default articulation) and Toggle — all of which might be useful in different circumstances.

As a one-time-only task, Articulation Map creation is not such a big deal if you regularly write with just a few of your favourite multi-articulation orchestral patches. However, if you have a number of libraries you use for different styles of writing, it would obviously be a bigger undertaking. Usefully, you can import pre-existing articulation maps and, interestingly, as well as those created specifically for use with DP, the import process also supports those created for the similar articulation map feature within Cubase (Steinberg call these Expression Maps). I tried this with some free-to-download articulation maps from Steinberg's website and the import process worked smoothly but I still found myself doing some customisation to suit my own preferences. That said, if you have some favourite multi-articulation tools you like to use, an online search might save you some initial setup time with DP's new Articulation Map system. Still, I'm sure hard-core DP users who do a lot of orchestral work will feel any initial configuration time is effort well spent.

Express Yourself

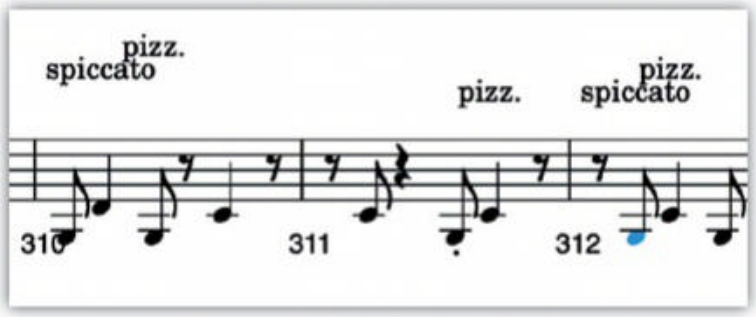
In line with a number of the other major DAW/sequencers, MOTU have now added MIDI Polyphonic Expression (MPE) to DP11. While MPE has now been around for long enough to feel like more than an interesting experiment, it is perhaps still something of a minority activity. That said, the creative potential is obvious for those willing to invest in a suitable controller (such as the ROLI Seaboard) and learn how to fully exploit it.

If MPE is your thing, then it certainly looks like MOTU's first iteration of it within DP11 is whole-hearted. Impressively, this includes making a number of DP's bundled instruments — Bassline, Modulo, MX4, Nanosampler, Polysynth and Proton — MPE-compatible. If you have a suitable controller, there are some steps required to configure your MIDI track to use MPE, but the MIDI data can then be captured on a single MIDI track. Editing is also well-implemented. For example, the per-note data is visible in the Sequence window and can be edited in per-note expression data lanes using the same tools as for conventional MIDI CC data editing. Incidentally, this includes the 'new for v.11' MIDI editing Scale Tool, which can be used with conventional MIDI CC data or per-note expression data. There are also some useful commands for managing MPE tracks/data. Overall, it's an impressively neat system given the potential complexity of the data that can be created.

More Live

One of the highlights (and surprises) of the DP10 release was the introduction of the Clips system aimed at live performance with DP. The live context is obviously something MOTU are keen to encourage as v11 sees some further developments in this area. While there are some refinements within the Clips Window itself, two technical changes are particularly worth highlighting.

First, DP now offers a Live Performance Mode that can be toggled on via the main Setup menu. This mode disables DP's PreGen engine that, by default, works in the background to pre-generate audio from some effects and virtual instruments. In a studio context, the PreGen system is a very good thing. It reduces the real-time load DP places on the computer host, ensuring there are resources available to keep any virtual instruments and/or effects being used in real time (for example, as you record a new part/track) responsive. However, pre-rendering may not be practical in a live performance context where you might, for example, be triggering clips and applying hands-on control on effects. Live Performance Mode ensures that everything is processed in real time and, while this might mean a heavier load on the host, within your CPU limitations, it should increase responsiveness of the system overall.

Second, DP11's Clips Window now includes integration for popular pad controllers such as Novation's Launchpad Pro and the Akai APC-40. These can therefore be configured to trigger individual Clips or Scenes (pre-defined

Articulation Map labels are automatically placed within the QuickScribe display.

collections of Clips) making for a much more tactile experience to live jamming with DP.

Hands On

And talking of hands-on control, MOTU has added support for a whole raft of newer control surfaces and made improvements for some older ones. The current range of NI's Komplete Kontrol series are now supported, along with the new iCON Control mixing worksurfaces such as the QCon Pro G2 and more budget-friendly models within the Platform/Nano series.

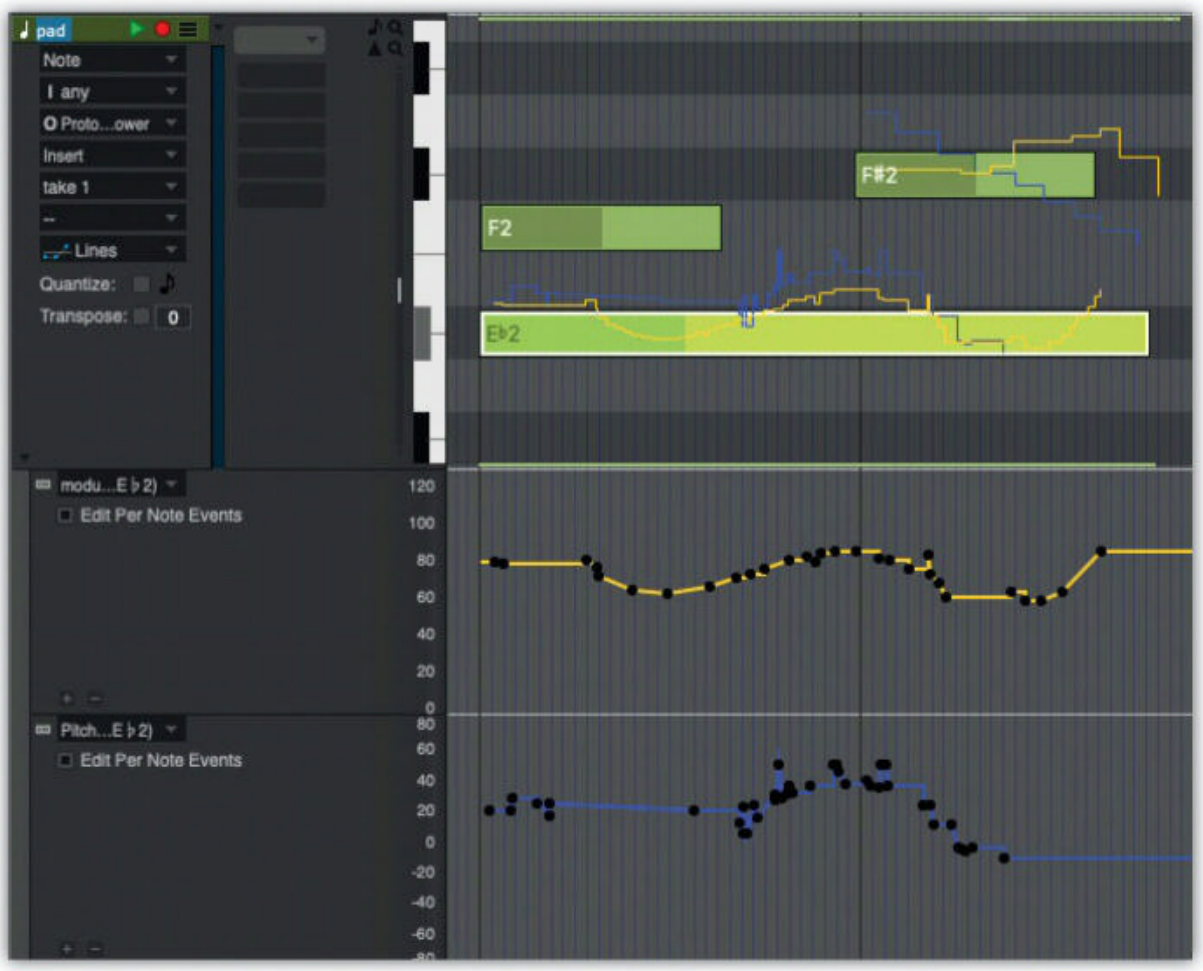
DP's support for the AVID EuCon ethernet-based control surfaces has also been improved and now includes the mighty Avid S6 as well as support under Windows 10. In addition, for suitable control surfaces, DP's V-Racks can be displayed. DP11 also offers improved integration with Softube's Console 1 devices and for devices that are Mackie Control Universal compatible. With external control surfaces now available to suit almost any budget, these additions and refinements will be broadly welcomed.

Best Of The Rest

While the selection of new and improved features detailed above perhaps represent the most obvious highlights of the v11 release, it's far from an exhaustive list. For example, for Mac OS users, it's good to know that DP11 is compatible with both Big Sur and Apple silicon, where DP's audio engine is optimised to make full use of the new CPU hardware. For Windows, DP's display customisation options now include the ability to resize text and produces crisper text for improved legibility.

The Track Selector has a new Channel Selector section that allows you to show/hide MIDI channel data and can be used alongside the Event Type Selector to show/hide specific event types. This kind of display filtering makes it much easier to navigate complex projects and focus on just the elements required at each editing/mixing stage.

Options for managing Chunks (DP's system for organising collections of tracks, songs, or collections of virtual instruments within a V-rack) have also been improved. Via the Chunks Window, you can now organise Chunks into folders and playlists and the Chunks list can now be split into two sections making it easy to search for an item in one section before copying it to a folder or playlist in the other.



DP11's support for MPE includes dedicated per-note expression data lanes for editing.

Heaven With 11?

Updating your favourite DAW/sequencer is always a mix of excitement (for the new features) and trepidation (will it run smoothly?). In terms of the latter, while I'm not sure the review process produced the kinds of stress-testing that a large project with a short deadline might have done, I have to say DP11 performed very smoothly throughout. If my own experience is typical, then existing users should (thankfully) find moving from v10 to v11 a relatively painless process.

Like most of the top-end DAW/sequencers, DP was already a very mature application and, with a loyal user base not wanting to see their well-established workflows disrupted, upgrades via evolution are really the only viable way forward. The DP11 upgrade does just that; evolution rather than revolution. Existing users will obviously have to weigh up the cost/benefits of moving to v11 based upon how useful they feel the collection of new and improved features are likely to be for their own workflows. For media composers — who form a healthy proportion of DP's users — I suspect the combination of Nanosampler 2.0, Articulation Maps and better control surface integration may be enough, while the 'always on' audio recording will also have a very broad appeal.

I'm perhaps less convinced that this release is likely to tempt a generally content Logic Pro, Cubase Pro, Pro Tools or Studio One user to suddenly jump ship. That said, if you are in the process of DAW divorce, DP11 should certainly be on your 'DAW to date' list. Yes, I would still suggest that you hang on to your existing third-party virtual instruments but, in all other respects, DP11 is a deep and powerful platform. Of course, that depth and power does come with the obvious qualifier of an initial learning curve but MOTU's flagship music production software is capable of handling even the most complex of projects. Turned up loud and proud to 11, DP11 remains an impressive DAW/sequencer beast.

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INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

Ari Morris

Hip-hop and metal share more in common than you might think, as mixer Ari Morris confirmed when he worked on MoneyBagg Yo's hit album, *A Gangsta's Pain*.

"I'm not a technical mixer. For me it's all about feel and vibe. It's all heart for me. There's very little science involved in any of my process. This means that I have a hard time explaining what I'm doing, because I just kind of close my eyes, and go. It's why I work a lot with rap music, because the beauty of rap music, and what drew me into it, is that it is all about vibe.

"Rap is not about things needing to be technically right. A 12 year old can make a beat at a high level, because it's all about a feeling. And for me, the sledge hammer feel of an 808 is the same as the sledge hammer feel of a big, chunky, heavy metal guitar. At the same time, you have to make each rapper sound distinct and uniquely them, to their core, so that if you're a fan, you'll instantly recognise them and are going to love their music."

Ari Morris may claim to have a hard time explaining what he does, but in just a few sentences he neatly summed up some of the principles at the heart of his process. Morris has put his vibe skills to good use with a range of rap artists, like Young Dolph, Future, Rod Wave, Royce da 5'9", EST Gee, 42 Dugg and, most of all, Moneybagg Yo. His vocal engineering and mixing credits on Royce da 5'9"'s album *The Allegory* (2020) earned Morris his first Grammy Award nomination (for Rap Album of the Year), and his ongoing mix work for Moneybagg Yo resulted in a US number-one album, *A Gangsta's Pain* (2020). The album spawned several big singles, including the platinum 'Time Today', and 'Wockesha', which reached number one on the Main R&B and Hip-Hop chart and the US Urban Radio chart.

'Wockesha'

Written by DeMario DeWayne White Jr, Mark DeBarge, Etterlene Jordan, Jorres Nelson, Christopher Pearson & Javar Rockamore
Produced by Real Red, YC & Rockamore

According to Morris, DeMario DeWayne White Jr, aka Moneybagg Yo, comes to him exactly because of their resonance over vibe, and the mixer's capacity to make the rapper sound unique. "He knows that I'm going to care about every second of his music just as much as he does. From there it's about the subtleties of how he likes his voice to be heard, and how he likes to hear music in general. He calls me because I understand the vision in his head, without the need to reference other things. One of the most beautiful things about working with an artist like Bagg with such a strong sense of self is that we never reference anyone else's music."

From Hardcore To Hip-hop

At this point it's necessary to take a few steps back, and examine where Morris himself is coming from, both literally and figuratively, and explain how he arrived at his vibe-over-technology approach, his relationship with Moneybagg, hearing 808s like heavy metal guitars, and ended up in Memphis. Morris takes up the story.

"I'm originally from New Jersey, and I started played guitar when I was 10 or 11. Most of my teenage years were spent playing in bands, and it's how I discovered recording. I started on the blue Tascam DP01 8-track, which was all digital. My first foray into making money from music was recording people's audition tapes for music schools with that machine! The hardcore punk movement was big in New Jersey, so I grew up as a metal head. I also was heavily into East Coast hip-hop.

"I came down to Memphis on a whim. I went to study at the Rudy E Scheidt School of Music at the University of Memphis, and found the huge musical heritage in this city, going back historically to Stax Records, Sun Studios and Royal Studios, and moving forwards into hip-hop and trap. I had been aware of the history, but not of everything that's happening today. There's a real music vibe here. Many people have innate, crazy artistic talents!

"While studying, I started working as an intern in studios in Memphis, and worked my way up. During that time I linked with a couple of artists who were coming up at that time in Memphis, like Young Dolph. I worked with him and his label Paper Route Empire for years. From there I linked up with Moneybagg, early in his career in 2016, and did a bunch of early projects with him that led to him signing with a major label."

Mentors

Morris mentions engineer/mixer Malcolm Springer and in particular Nil Jones as his mentors. "Nil is very into soul, funk and hip-hop, and known for the »

Ari Morris.

» implementation of a live band within trap music. I sat behind him mixing for years, and I learned everything from him! I learned from him how to be in the studio, which is the first thing, and how to translate an artist's vision into their record, rather than your own. It always has to be about the artist and what they're looking for. I also worked in studios in Atlanta and LA, until about five years ago, when I settled into mixing in my own comfy studio place in Memphis."

So, how did a self-declared 'metal head' end up working predominantly in rap in general and trap in particular? "I've always loved metal and rap, but when I came down here and really started experiencing the trap community, it felt very similar to the punk and hardcore community. It's the same ethos, the same attitude: 'Fuck you, I'm going to do what I want to do.' It's that mentality. It is so raw. It also is a raw recording style. With the early trap records that I mixed, in my head it was heavy metal with 808s. It was like, how do I channel this raw aggression into this record?"

It was their mutual love of channelling raw aggression in recording that drew Moneybagg and Morris together in 2016, and that led the rapper and his label in 2019 to call Morris back in, after the rapper's first two albums on Interscope were mixed by others. "I did engineering and mix work on some of his early mixtapes like *ELO* and *All Gas No Brakes*, and the *2 Federal* mixtape with Yo Gotti that led to the label deal (all 2016). There was a vibe and an energy in the vocals on those, and when they came back to me they said they were after that vibe.

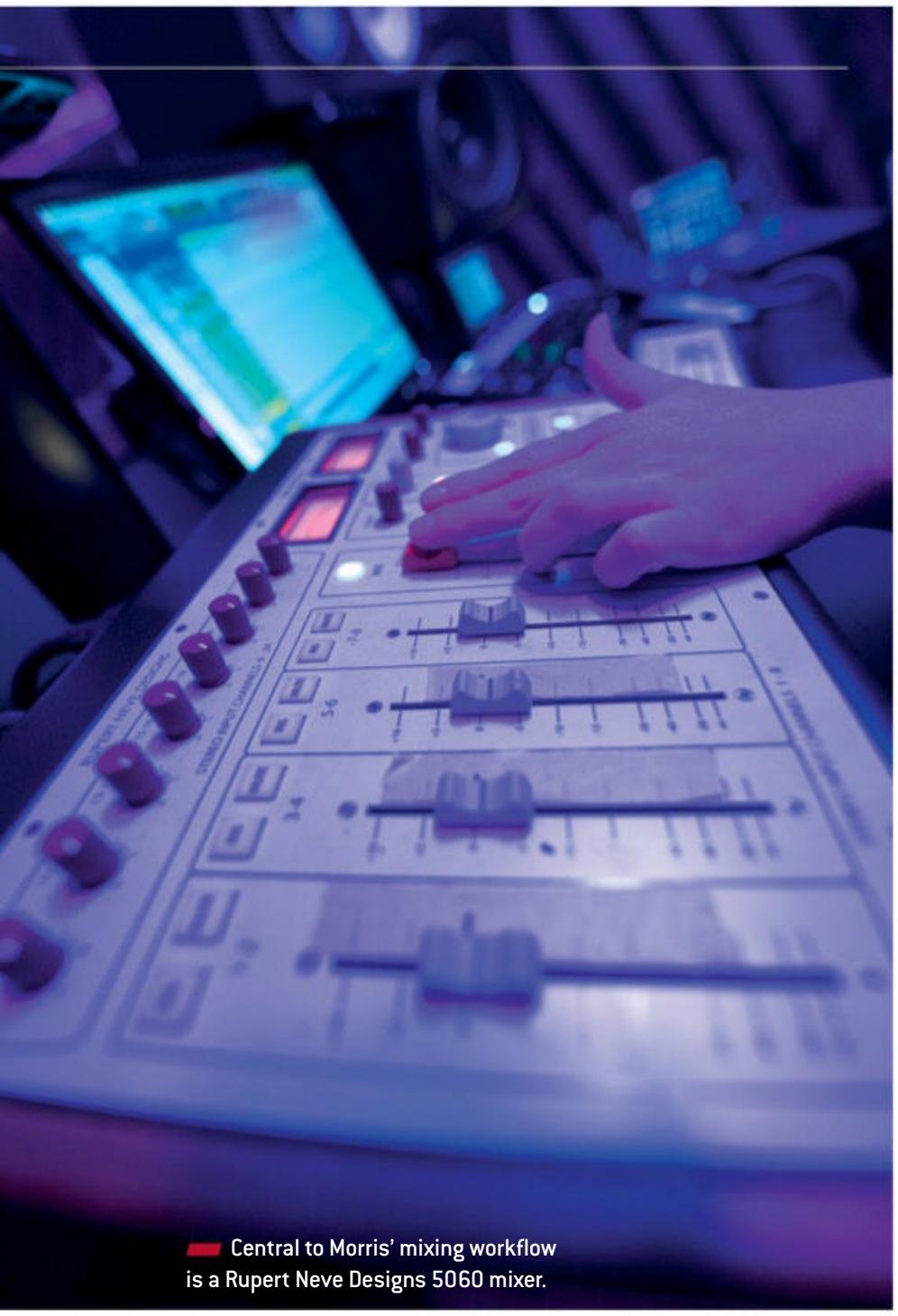
"So when I started working with Bagg again in 2019, I went for this vibe, which sits in the tonality of his vocal and in the placement. I put him in a different spot than where many other people would place a vocalist. It's unique and distinct. It's about creating something that's in your face and bone-dry. Bagg hates the traditional ear candy like delay throws, reverb swells, reversing things and panning them around, and so on. It's not how he feels music. It takes you out of the moment. Instead the feel with him is bone-dry, raw power, maintain the size and the weight and the heft of a real record, and create this big thing that's not covered in fluttery ear candy."

Winding Up

Another essential ingredient in the unique sound Morris gets is, he says, "transformers, transformers, transformers". Which brings us to his studio in Memphis, where Morris has mixed everything Moneybagg Yo has released over the least two years, including the albums *Time Served* (2020) and *A Gangsta's Pain*. For four and a half years Morris worked in a tiny 8 x 10-foot control room, but he recently moved to a much larger space, which is set in a 3000 square-foot warehouse.

"*A Gangsta's Pain* was the last big project that I mixed in my previous place. But both my studios have the same gear. All I added when I came to my new place was a McDSP APB. I love that box. My monitors are Amphion One18s and Two18s, and Genelec 1032As, which are 28 years old, paired with a BassBoss DJ18s subwoofer. I used to be an NS10 guy religiously, until I heard the One18, and I was like 'Oh, shit,' swapped my NS10s for those, and never looked back. My I/O for Pro Tools is an Apollo 8, which feeds my Neve 5060, and Burl B2 Bomber A-D/D-A for printing back into Pro Tools.

"I have a hybrid setup, and the Neve 5060 Centerpiece desktop mixer is the heart of my studio. It also functions as my monitor controller. It has four input faders and a mix fader, and no EQ



Central to Morris' mixing workflow is a Rupert Neve Designs 5060 mixer.

or compressors, just in and out transformers on every channel. I was trying to find a monitor control that sounds like a console, because I came up working in rooms with a Neve or an SSL desk. The main desk I worked on was a Neve VR60, and it is my favourite board to this day. The 5060 gives me a similar sound. It's just channels of irons. Lots of transformers.

"In addition to the 5060, I have a couple of key compressors, the UA Rev D 1176 and a Rev A 1176, for if I really want to pump the vocals. I also have Vintech X73i and Avalon VT737 mic pres, and the output of my 5060 goes to my AudioScape G Buss Compressor, and then into the Burl B2 Bomber, which is another set of transformers. Lots of transformers and stages in analogue that can clip and be hit hard are my answer to getting low mids in my records."

The latter remark is a reference to a problem that many mix engineers today face: with the focus on heavy low end and sub bass, and extreme brightness, the midrange is often forgotten. This despite the fact that, as Morris acknowledges, "Your money is in the midrange. Getting the midrange in there and making sure it feels like a record is part of the art of being a mix engineer. For me the answer is lots of saturation from plug-ins, for example using iZotope's Ozone, Soundtheory's Gulfoss, the SPL IRON compressor, and then all the transformers in my studio."

Stems Or A Two-track?

Morris was asked to illustrate his mix approach with his mix of Moneybagg Yo's track 'Wockesha', the most recent hit from *A Gangsta's Pain*. The mix engineer's reaction was »

Morris mixing at his “comfy new studio” in Memphis, with his two woofers.

» illustrative of his general vibe-over-tech attitude. Over the last few years it’s become a common complaint among mix engineers working in the trap genre that they are sometimes just required to mix a handful of vocal tracks in with a two-track (stereo print) of the beat. To say that top mix engineers are over-qualified for these mixes is an understatement.

It turned out that the ‘Wockesha’ mix session is a case in point, as it also is extremely simple. The beat is built around a sample of the 1983 DeBarge song ‘Stay With Me’, reinforced by producers Real Red, YC and Javar Rockamore with just four drum elements. With six vocal tracks, the mix session has only 11 audio tracks, not counting a 20-second sample of Lil Wayne talking, and a couple of very short sound effects. But where some mix engineers rue that they cannot use more of their skills in such simple sessions, Morris takes a different view.

“Yeah, this song is an immensely simple record. But aren’t the biggest records sometimes not the simplest? It’s the ones that don’t need all the extra shit that are the ones that cut through, because they really are good. With this record, the vibe was there, so it didn’t need a lot. When I received the session to mix, the track was already in a really special place, and Bagg was adamant that he liked the way the beat sat. So why do a lot?

Ari Morris: I’ve always loved metal and rap, but when I came down here and really started experiencing the trap community, it felt very similar to the punk and hardcore community. It’s the same ethos, the same attitude...”

“Generally, with all mixes I do, I get the track-outs [*stems of individual tracks*]. But also in that case you need to honour somebody’s rough mix. It’s the name of the game now. People spend months sometimes on their rough mixes. There’s an intent there, and you need to enhance that intent. If something is really simple, you can say you’re over-qualified, but really, there’s also an art to enhancing someone’s idea with a really subtle hand, without losing their vision.

“One of the things about a beat like this, where there are only five elements in the beat — sample, 808, kick, snare and hat — is that if you change the sonics of any of these, everything changes, and it’s a whole new piece of music. The minute you change anything, other than fixing something obvious, it falls apart. It’s not like when you’re working with a piece of music with 30 elements, and you can change the character of something, and nobody notices. So with a mix session like this, it’s all about being incredibly subtle. Mainly what I did was pulling in some midrange, using saturation and transformers, and mostly processing things together, because it means you’re adding an overall glue.”

Session View

Morris’ mix session for ‘Wockesha’ consists of 31 tracks. From top to bottom there are: the rough mix (REF), the two-track of the beat, the five drum tracks (808, snare, clap, hat, sample), the Lil Wayne spoken word sample, the two sound effects samples, a submix track for the beat, six vocal audio tracks, a submix track for the vocals (titled ****), a submix track for the backing vocals (confusingly called ADLIB), a Doubler aux track with the Waves Doubler 2, two more vocal

Morris' outboard rack, featuring (from top) an Avalon VT737, a Universal Audio 1176 Rev D, a Vintech X73i input channel, a dbx 160X compressor, and an 1176 Rev A.

submix tracks, five aux effect tracks (two with reverbs and three with delays), Morris' return track from his 5060, two mix print tracks and in between them a master track.

Although the aux tracks are arguably part of a template of sorts, Morris says he's not keen on using templates. "I prefer to start with something like a blank canvas, other than that there's some routing with a few key tools in there, but almost all plug-ins are at factory default settings. Once I go into album mode, then yes, if the vocals for that album are all recorded in the same spot or similarly, I'll pull similar vocal treatments across different mix sessions.

"Thankfully, with this album Moneybagg's mix engineer cut everything with a Sony C800G, so there was a level of consistency. A lot of what I'm doing in other situations is bridging the differences between vocals recorded across multiple locations, multiple days or months or even years sometimes. You have to get creative to make those fit together.

"With Moneybagg I receive the Pro Tools session, and my assistant preps it, which is mostly a matter of colour coding and labelling stuff, so I know what I'm dealing with, and it looks the way I want my sessions to look. I can then just sit down and start mixing. I'll listen to the rough a couple of times, and then I start running with it. I normally touch the music for a bit, then go to the vocals and work on them for a while, come back to the music, and back to the vocals.

"Once it gets to something that I think sounds good, I take a break, and when I come back I see where I'm at. Sometimes this means that I'll pull everything down and start again. I learned to mix on analogue consoles,

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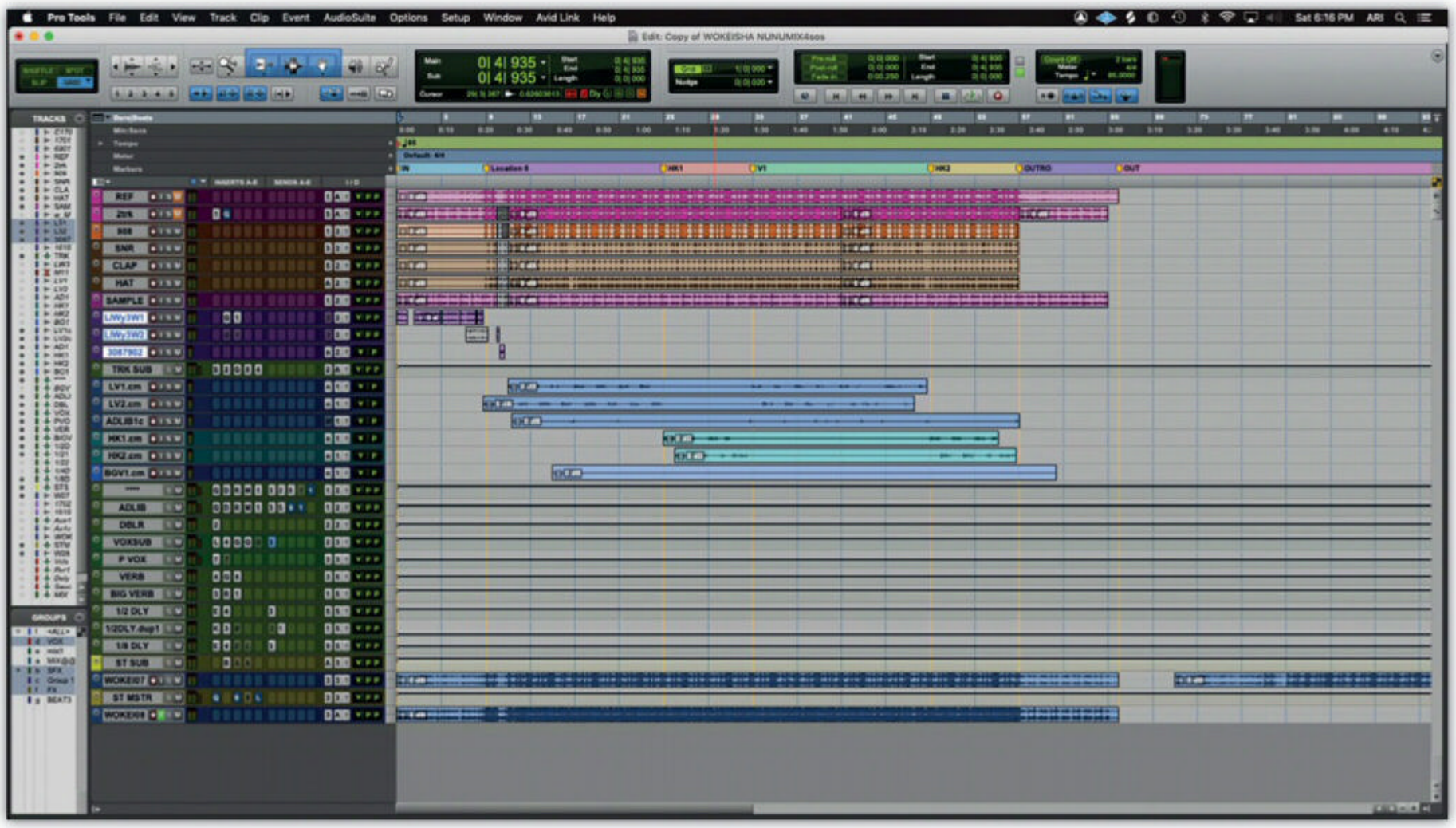

The Session view for Ari Morris' mix of 'Wockesha'.

» and a studio manager once said to me that a mix captures a moment in time. When you listen back later, sometimes it's right, and sometimes it's not. You stay in the moment, and don't get mired in second-guessing yourself. I'll finish my thought, my moment, and there's a risk that I sometimes later realise that I don't like it."

The Beat

Morris already explained above that he wanted his treatments of the beat to be very subtle, while 'pulling in' some midrange with saturation. For the beat in this session he does this on the Track Sub aux track.

"My signal chain on that track starts with parallel compression from the SPL IRON, hitting the Germanium side to add more weight. Next is the Oeksound Soothe. I love that plug-in. It's a genius piece. It just kinda softly curls under a couple of frequencies that I thought were a little too prominent, by probably half a dB. Then there's the Gulfoss, which is another plug-in that I like a lot for that low-mid area. Next is the Ozone 8 Exciter, again bringing in some low mids, and the Mäag EQ4 has a sub-bass boost and some high-end air. The Kush Clariphonic EQ is doing a similar thing, putting some more shimmer on what's happening.

"Working with various types of saturation is like putting a magnifying glass on certain frequencies, rather than straight adding them with EQ. At the same time, these EQs are very adaptable, and don't feel as phasey to my ear as some regular EQs do. They don't sit statically across the beat, but move with it, allowing me to be very subtle, in a way

Among the few new additions to Morris' Memphis studio was the McDSP APB. Below it are two Burl B2 transformer-balanced converters, through which Morris prints his mixes.

Mix Routine

Mixing *A Gangsta's Pain* took Morris six months in total, off and on, and six weeks of continuous work at the end. Given that Morris says that mixing trap is all about vibe, how does he make sure that what gives him a particular vibe on one day gives him the same feeling on other days? "When I'm really in crunch time for an album, I'm like an athlete in the play-offs," says Morris. "There are no haircuts, there's no shaving, and there's waking up, and breakfast and going to bed at the same time every day. I try to stick to the same daily routines every day, so it may be, 'OK, we did the poke bowl from the grocery store today for lunch, I guess we'll be doing it all week!' As much as I can I control my environment and myself, so I'm very careful when I answer phone calls, and from who, and what energies I allow into my life during that period. Mixing is a game of mental fortitude, so I do what I can to stay in the same zone for the whole duration of a project."

that nobody really notices. People just go, 'Oh, it's a little clearer and a little richer.'"

Vocals

The vocal group track, to which all vocal audio tracks are sent, and the AdLib aux to which the backing vocal audio track is sent, have very elaborate and almost identical vocal chains, consisting of FabFilter Pro-Q 3, FabFilter DeEsser, Waves RCompressor, UAD Manley Massive Passive EQ, UAD dbx 160 compressor, UAD Thermionic Culture Vulture, and, in the case of the main vocal group, the FabFilter Pro-MB and the Soothe 2, plus several aux sends.

"I used a few different vocal chains on this album, depending on the song and where the vocals were recorded. But in general it's the same idea, with the Q 3 pulling out unwanted resonances, and the DeEsser just grabbing things at the top. The attack on the RComp is faster on the AdLib aux, to create a different character. I love the Manley, because you can add midrange without things going wonky. The 160 really crushes the vocals, something that I used to love doing on the hardware.

"The Culture Vulture is adding more weight and Pro-MB is dipping some low end. With a husky voice like Bagg's, it's a delicate balance where his low end sits against the low end of the beat. There are also sends to the Doubler, and the reverb auxes, but on this song, I use them very minimally. Both vocal aux tracks are sent to the Vox Sub, which has the FabFilter Pro-L 2, just kissing the signal a bit, a Mäag EQ4 for more air wizardry, and the Quartet DynPEQ also does great stuff on the top end. I parallel compress the Vox Sub, with the Waves 1176 on the P Vox track next to it.

"The Vox Sub and P Vox tracks, as well as all my aux effect tracks, are bussed out to the Neve 5060. Everything gets slammed in there, and again by my AudioScape Buss Compressor, and via the Burl comes back to the Stereo Sub aux. The Burl has a stepped input transformer, which is nice for pushing stuff extra hard on coming back. It soft-limits in a really nice way. The Stereo Sub track has the Black Box Analog Design HG-2, again for more saturation, and I print the stereo mix next to it twice, once for the mastering engineer, and a second version after treatment by the Pro-L 2, to push the volume up for my clients."

In the end, Morris did get quite technical. But it's the vibe and feel of his work that speak the loudest.

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Immersive audio has huge creative potential for music production, but it can be hard to get your head around. Here's the explanation you've been waiting for!

SAM INGLIS

From long-playing vinyl, eight-track cartridge and compact cassette through to CD, Minidisc and MP3, most consumer audio formats have had one thing in common: stereo. They contain two discrete audio signals, designed to be played back through two loudspeakers or earpieces. A listener positioned between these speakers hears a 'sound stage' or panorama, within which individual sources appear at specific positions. For example, a mono vocal at equal levels in both channels will be heard as being halfway between the two speakers, directly in the centre.

This illusion is impressive, but it's also limited. It lets us localise sources along a line between the two speakers, and as being near or far away, but it can't convey a sense of height, or reliably convince us that sound is coming from behind us.

Over the years, there have been several attempts to overcome these limitations, most notably quadraphonic sound in the '70s, Dolby Stereo (ProLogic) through the '80s and '90s, and 5.1 surround in the early part of this century. However, these achieved lasting success only in the cinema. Despite considerable investment from record companies, domestic audiences didn't warm to the new formats.

There were several reasons for this. Marketed as premium products, quadraphonic records, DVD-Audio discs and multichannel SACDs were more expensive than the stereo versions of the same material. They could not easily be enjoyed on headphones, and required specialised playback equipment, including at least four loudspeakers. This, again, was costly, and it was impractical or at least undesirable in many home environments. Even if you had the money, the space and the domestic goodwill to set up a 5.1 speaker system, moreover, the benefits would be confined to a very small 'sweet spot'.

Breaking The Link

What stereo, quad and 5.1 all have in common is that they are channel-based formats, meaning that there is a fixed relationship between channel count and speaker count. Each discrete channel carries a signal that's destined for a specific loudspeaker, and the loudspeakers themselves need to be configured in a specific physical relationship. In the right space, with everything set up correctly, the experience could be magical; but in practice, such spaces and setups were few and far between.

Ambisonics is a capture format as well as a replay format, and Ambisonic mics such as the Rode NT-SF1 can be used to record very convincing surround ambiences.

There are two key features of modern immersive audio formats. One is that they don't just represent surround in the horizontal plane: they also contain meaningful height information that allows sounds to be perceived as being above the listener. The other is that nearly all of them break the simple relationship between channels and speakers. The delivery format does not simply contain a separate mono channel destined for each speaker, but a more complex data stream that is decoded in real time to map it onto whatever speakers are available in whatever locations. Unlike older surround formats, immersive audio therefore requires an 'intelligent' device in the replay chain to carry out this decoding and customised mapping. But in many contexts, this isn't really a problem, because the stream is being played back from a computer, server or other device that has plenty of spare processing power.

Certainly, the possible need for an additional device in the signal chain is massively outweighed by the benefits. The key plus is that, in principle, immersive content can be decoded for any speaker arrangement you like, from the bandwidth-limited mono speaker in a smartphone to a full cinema array with many rear and side speakers, overhead speakers and subwoofers. It can also be fed through a binaural encoder to achieve a sense of immersion on headphones. In other words, whereas previous surround formats required listeners to adapt their setups to suit the format, immersive audio adapts itself to suit whatever setup is available.

We can classify immersive audio formats as being channel-based, scene-based or object-based. The first category refers to surround formats beyond 5.1 that incorporate speakers above the listener, and can therefore claim to be immersive whilst retaining the simple direct and »

There is currently a fierce format war going on in the world of immersive audio for music. Thanks to its adoption by Apple and others, Dolby Atmos looks to be in pole position at the time of writing, but it's too soon to write off any of its rival contenders.

» exclusive channel-to-speaker mapping. Scene-based formats, by contrast, present a single, complex data stream that describes a complete three-dimensional soundfield. Finally, object-based formats package a number of discrete audio streams along with metadata that tells the decoder how these individual streams should be positioned. Crudely put, channel- and scene-based formats contain fully mixed audio, whereas an object-based format contains the major elements of a mix plus metadata explaining how that mix should be implemented in a given playback environment. At the time of writing, there are a number of commercial audio production and distribution formats that are billed as immersive, spatial or 3D, all competing to dominate various market sectors. As we'll see, many of these are in fact hybrid formats that combine object-based elements with channel- or scene-based elements.

The Scenic Route

The main example of a pure scene-based 3D audio format is Ambisonics. Developed as long ago as the late 1970s by Michael Gerzon and Peter Craven, Ambisonics can be thought of as an extension of Mid-Sides stereo. Mid-Sides is also known as 'sum and difference', and first-order Ambisonics extends the concept by adding two additional 'difference' channels, representing the front-back and up-down axes. The sum or W channel describes the omnidirectional component, while the subsequent X, Y and Z channels describe the directional components of the sound in the three orthogonal planes.

Ambisonics can be scaled to an indefinite number of orders. The number of channels needed to implement a given order n is (n+1) squared, so second-order Ambisonics requires nine channels, third-order 16 and so on. The advantage of higher orders is that the listener is able to localise sources more precisely within the soundfield. For example, consider two different sound sources starting at the same point and slowly moving apart. The higher the order, the narrower an angle we can discriminate between them.

There is no real-world playback system that can play back a raw Ambisonics signal. So, just as Mid-Sides stereo needs to be matrixed into left and right channels for playback on conventional loudspeakers or headphones, an Ambisonics signal must be processed to adapt it to whatever playback system is available. This can involve extracting separate mono signals for each speaker in a surround array, or running the signal through a binaural encoder for headphone listening.

Ambisonics is thus similar to stereo and 5.1 surround in the sense that each decoded channel describes part of a complete soundfield, rather than a discrete element within that soundfield. Where it differs from those formats is that the one-to-one relationship between channels and loudspeakers is broken. An Ambisonic signal is an abstract representation of the soundfield that needs to be matrixed to a particular listening environment.

Objects Of Desire

In an object-based format, by contrast, each channel describes a specific element of the mix: a vocal, an instrument or group of instruments, a Foley effect such as an explosion, or whatever. Each object is packaged with its own set of timecoded metadata, which is derived from and often identical to mix automation data. For example, the metadata for a percussion track might tell it to remain in the distance at the upper left of the soundfield until 30 seconds into the track, then move slowly across the top of the listener's head and get closer before performing a quick pirouette. To deploy a rather leftfield analogy, a scene-based audio mix is like a fully baked loaf that only needs to be sliced up, while a channel-based mix is a pre-sliced loaf — but an object-based mix is like a part-cooked loaf that comes with instructions as to how it should be finished off.

Scene-based and object-based formats can both deliver very impressive results, but in their pure form, they have different strengths. Higher-order Ambisonics excels at presenting natural-sounding ambiences: if you want to create the impression that the listener is really standing in the middle of a jungle or a busy city street, this can be achieved very effectively. This is partly because Ambisonics isn't just a replay format. It's also a capture format, and the use of an Ambisonic microphone such as the Soundfield permits direct, natural recording of a three-dimensional soundfield.

The flip side of this is that Ambisonics can feel a little understated, and even in second and third orders, offers relatively limited localisation. An Ambisonic presentation of a soundfield is analogous to a coincident stereo recording, in that all of the directional information on offer derives solely from level and tone differences, and not from time-of-arrival differences. Those who have worked with it say that it's hard to make a single instrument or mix element really leap out at the listener, or to create a really precise sense of its occupying a specific position.

This is where object-based formats come into their own. Because the positioning of objects is implemented locally and with reference to the particular »

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The Dolby Atmos Renderer is a standalone piece of software that handles object panning, routing, monitor control and ADM authoring.

» speaker layout in use, they can be placed and moved with a precision and 'in your face' quality that is otherwise difficult to achieve. Rock and pop mixing is often more about power, drama and immediacy than it is about naturalism, and this sort of larger-than-life presentation is easier to achieve with object-based mixing than through Ambisonics.

The Best Of Both Worlds

Although data bandwidth and storage space are much less pressing issues than they used to be, both are still finite resources. If every single source within a mix is treated as an individual object, channel counts and file sizes rise alarmingly, especially in complex soundtrack-type music, or where music coexists with dialogue and sound effects. Yet in most music mixes, there are quite a few elements that don't need the precise localisation or detailed positional control that objects offer; and a workflow that forces the mixer to assign every single instrument in a busy mix to its own object would risk being very slow and inefficient.

Consequently, many of the immersive formats that are currently jostling for space in the market have a hybrid nature, combining object-based playback for important mix elements with one or more scene- or channel-based streams that can mop up the rest. Dolby Atmos is a good example: in addition to its objects, an Atmos mix can contain one or more 'beds'. Each bed is, in essence, a conventional channel-based stream in up to 7.1.2 surround (ie. seven horizontal channels, one LFE channel, and two height channels); when the mix is played back on a system with fewer speakers, it gets intelligently folded down to 5.1, quad, or whatever is appropriate. This has the added benefit that older surround mixes are more or less directly Atmos-compatible, since they can be represented as a simple Atmos stream containing only one bed and no objects.

A Dolby Atmos mix can contain up to 128 discrete mono audio channels in total, divided between beds and objects. It would be unusual to use more than one bed in a music mix, so assuming that's a single 7.1.2 bed, you'd be left with a maximum of 118 mono or 59 stereo objects available. The rival DTS:X standard is similar to Atmos in that beds are augmented by objects, but in this case there's no upper limit on the number of objects. Sony's 360 Reality Audio is built on the MPEG-H 3D Audio standard, a more open-ended format that can contain objects, channel-based audio and Ambisonics data. Auro Technologies' Auro 3D, meanwhile, began as a purely channel-based system, but the most recent AuroMax iteration has added objects.

Immersive Mixing

With stereo and channel-based surround, individual channels within a mixer are routed directly or indirectly to a two-channel or 5.1 master bus. The outputs from this bus are then routed to individual speakers, and generating a master recording is simply a matter of capturing the output from this multichannel master bus.

In the immersive world, things get more complicated. The whole point of scene-based and object-based formats is to be agnostic about the replay format, so in essence, the appropriate collection of scenes or objects needs to be generated and then decoded in real time for monitoring. As a simple example, consider the creation of a third-order Ambisonics mix from a number of individual mono sources. Each of those sources must be routed through some sort of 3D panning device or algorithm which can address the 16-channel Ambisonics bus; and there needs to be a further decoding algorithm that can map the output of that bus onto whatever speaker array we happen to have. If we want to audition the results on headphones, yet another step is involved, as the output must be re-encoded binaurally.

Object-based audio adds a further level of complexity, because of the 'part baked' nature of the format. We're not simply recording a multichannel output file that consists only of audio data. Instead, or additionally, we are outputting tens or »

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» hundreds of individual mono and stereo streams, each with its own collection of metadata. Few DAWs are currently set up to produce this sort of output natively, so the usual approach is to integrate an additional piece of software that can receive all the necessary channels and data. In Atmos-speak, this device is called the Renderer, and it acts as both monitor controller and master recorder.

Turning a mix source into an object involves assigning it to its own unique mono or stereo path through this virtual device, and replacing the standard panner on the source channel with a surround panner. This panner sends instructions to the renderer where they are recorded as metadata, and interpreted by the monitor controller according to the appropriate speaker mapping. In other words, the panning and movement of objects within the DAW mixer is actually implemented or duplicated within the renderer. In Pro Tools and Nuendo, the standard channel surround panners are 'Atmos native' and can be switched on the fly to produce either object-based or channel-based metadata. Logic requires the channel itself to be set to channel or bed mode, with different panners used in each.

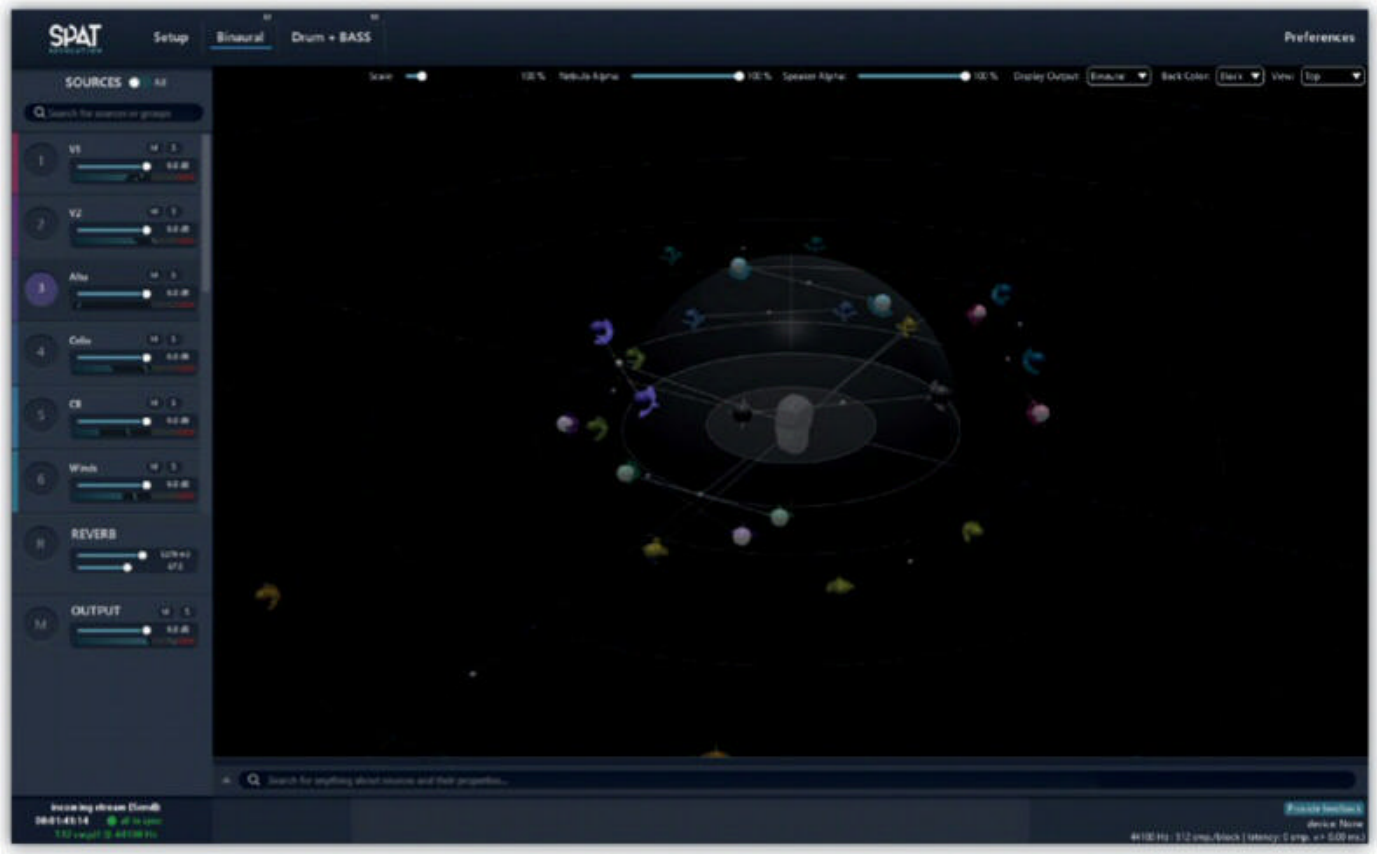
Integration between renderer and DAW can be handled in a number of different ways. Apple have recently integrated a streamlined version of the Dolby Atmos renderer within Logic, along with appropriate tools such as object panners, making it possible to create an Atmos mix and author the necessary ADM (Audio Definition Model) file without leaving the application. (In many cases, the ADM file is formatted as a Broadcast Wave, aka BWF, file for the audio objects, with a large data chunk added containing the associated metadata.) Steinberg's Nuendo also builds in a version of the Atmos renderer. For other DAWs, Dolby themselves make available two separate products called the Production Suite and the Mastering Suite. The cross-platform Mastering Suite is designed to run on a separate computer from the DAW, essentially combining the functions of master recorder and monitor controller; typically, you'd pipe audio from the DAW machine to the Mastering Suite machine using a high channel-count digital format such as MADI or Dante. The Mac-only Production Suite, by contrast, installs a virtual Core Audio soundcard called the Dolby Audio Bridge on the DAW machine. This in turn pipes audio to the renderer software, running on the same

Many engineers creating immersive mixes prefer to do so from a control surface such as Yamaha's Nuage or Avid's S-series.

machine, and handles monitor control and so on. Both approaches have their pros and cons.

The nature of object-based immersive audio means that some techniques common in stereo mixing need to be 'unlearned'. Many rock and pop mixers depend heavily on master bus processing to craft the sound of a track, but in an object-based immersive format, there is no master bus. Even in channel-based surround, processing the master bus can produce very different results, and the idea of using master dynamics and EQ to 'glue' a mix together is rarely applicable. A related concern is headroom and metering. One of the reasons for using master bus processing in stereo is to increase the perceived loudness, and as long as we don't attempt to exceed 0dBFS at the master bus, the resulting mixes will be compatible with all stereo playback systems. In an immersive format, by contrast, headroom is not only desirable but essential: what goes to each speaker in the listener's system is determined on the fly during playback, so there needs to be sufficient headroom to ensure that these calculations never produce an 'illegal' signal level. You'll typically be working to a target loudness measured in LUFS rather than to a peak level.

Finally, one of the fringe benefits of object-based immersive audio is that it facilitates archiving of projects and transfer between different DAWs. As mentioned above, the ADM format that is used for Atmos masters is built on the familiar Broadcast Wave format, and represents each object as a mono or stereo WAV file plus associated metadata. So, if you take an ADM file created using Pro Tools with the Dolby renderer, and open it in Logic



Flux Spat Revolution and Dear Reality dearVR are third-party packages that allow sources to be panned within a virtual 3D environment for binaural, Ambisonic or channel-based surround.

10.7, you should find all your objects and their panning information faithfully recreated there (as long as it only has one bed: Logic doesn't currently support multiple beds).

Down The Pan

Most DAWs don't offer the option to use third-party stereo panners, because the job of a stereo panner is so basic that there's not really any point. However, when we move from stereo to surround, and especially to object-based immersive surround, the panner becomes much more important. Moving sources in three dimensions raises many new questions: How should that movement be visualised using animation? How should gestures and mouse movements be translated into changes in position? How can complex 3D movement be represented and edited in two-dimensional automation lanes? Should 3D panning simply mean making sources louder over here and quieter over there, or should those changes also be reflected in the ambience of a virtual space? And how should the perceived size or width of a sound source vary as it is moved closer or further away?

Engineers have different preferences regarding all these issues and more, and so there's a wealth of third-party software available that can augment or replace the tools bundled with our DAWs. One example is the Flux IRCAM Tools Spat Revolution software. Much more than 'just' a panner, it's perhaps better thought of as a renderer that can position and move any number of sources within a synthesized acoustic, then generate an Ambisonics, channel-based or binaural output. And although it can't directly create object-based content, it could be integrated within an Atmos workflow, for example to handle the creation of 7.1.2 beds as part of an Atmos mix. The same goes for other products such as Dear Reality's dearVR.

Even if the ultimate destination format is going to be Dolby Atmos, it's possible to reach that destination using third-party tools. A good example here is L-Acoustics' L-ISA system. This is a suite of tools for working with 'pure' object-based immersive audio either in a live or a studio environment. It can be integrated with a DAW in much the same way as the Dolby Atmos Production Suite can, presenting itself as a virtual soundcard and handling speaker management, binaural encoding and so on; and further down the line, L-ISA objects can be transformed into Atmos objects if you want to render your immersive mix in a format that can be distributed through the usual consumer channels such as Apple Music. Unlike Atmos, L-ISA also features a Room Engine which allows objects to be positioned within a virtual acoustic space.

Mention of live sound introduces another relevant concern. Dolby Atmos and other formats that originated in movies are typically highly regulated. To be qualified for the production of music or cinematic content in Atmos format, a studio has to meet pretty stringent requirements, which must be certified by Dolby themselves; it must also be calibrated for working at a specific reference sound pressure level. In live sound, by contrast, it's unreasonable to expect venues and sound systems to exhibit the same uniformity. Creating a workable immersive experience in live or installation sound is about achieving what's possible in the space that you have to work in, which may have unfriendly acoustics, an awkward shape and an audience too large to fit into any sort of 'sweet spot'. Hence, systems such as L-ISA and Spat Revolution are designed to work with non-standard »

» speaker layouts as well as the 'approved' setups you find in studios and cinemas.

On Stream

As was previously mentioned, master files for immersive formats can be huge. In theory, for example, a Dolby Atmos ADM file can contain up to 128 channels of 48kHz, 24-bit audio, plus associated metadata. That's over 18MB per second, or 1.1GB per minute. Clearly, it's not feasible to stream uncompressed Atmos masters over domestic internet, cable or satellite TV, or phone networks. Some fairly radical data compression is required and, again, the exact approach taken is format-specific. Since we're considering music distribution here rather than broadcasting or movies, Atmos is again the most relevant example owing to its adoption in Apple Music.

There are actually three Atmos Delivery codecs that form part of the Dolby Music framework. TrueHD is used exclusively on Blu-Ray discs, while AC-4 IMS is a low bit-rate format designed specifically for streaming: it contains a bitstream that is compatible with conventional stereo devices plus binaural metadata that enables users with headphones to experience the immersive content. The third codec is known variously as Dolby Digital Plus with Atmos Content or Dolby Digital Plus JOC, where the last three letters stand for Joint Object Coding. In essence, this packages a data-compressed 5.1 surround stream that is compatible with legacy devices with an additional layer that permits object-based playback to be reconstructed for Atmos-compatible systems.

Apple's Logic 10.7 promises to democratise immersive audio through the full integration of Atmos tools at no extra cost.

Most of us are accustomed by now to knowing how our speaker-based stereo mixes will translate to headphone listening, but immersive content is a whole different ball game! Relatively few among the target audience will have Atmos-compatible speaker systems at home, so the vast majority of immersive listening takes place on headphones. It's therefore vital to audition the binaurally encoded output as part of an immersive mixing or mastering workflow, but even then there are pitfalls. For example, objects and beds within an Atmos mix can be tagged with distance metadata that is specific to binaural encoding. Dolby's own binaural encoder (as used, for example, by Tidal) responds to this metadata, but Apple Music uses a different encoding system which does not. This is problematic, because the same Atmos mix may sound different when auditioned through the Dolby binaural encoder and through Apple Music, but there's currently no way to audition a mix through the Apple Music processing in real time. The current workaround is to export a mix from the Dolby renderer in MP4 format, Airdrop it to an iPhone and then play it back over some Airpods, which is not ideal.

L-Acoustics' L-ISA is a system for generating object-based immersive audio, either for live sound or for distribution using a format such as Atmos. The company have recently launched a version called L-ISA Studio, which runs natively on a Mac.

The Future Is Bright

Both old and new surround formats have, until now, presented a fairly high barrier to entry. Many people who have been excited about the creative possibilities have been frustrated by the cost and difficulty of working in surround. As a result, immersive audio has been almost entirely the province of post-production until now. If record company budgets permit, artists and producers have been able to go into an Atmos-certified studio at the mix stage and create an Atmos or other immersive mix to release alongside the stereo mix, but they haven't been able to work immersively right from the beginning of a project.

With the release of Logic 10.7, that looks set to change. Logic is positioned primarily as a music creation and recording program, and the integration of Atmos tools — at no additional cost — means that anyone with a laptop and a pair of headphones can make music immersively. Music production promises to be very different when immersive listening is borne in mind at every stage of the creative process, and I've no doubt that other DAWs will take up the challenge very soon. We live in exciting times!

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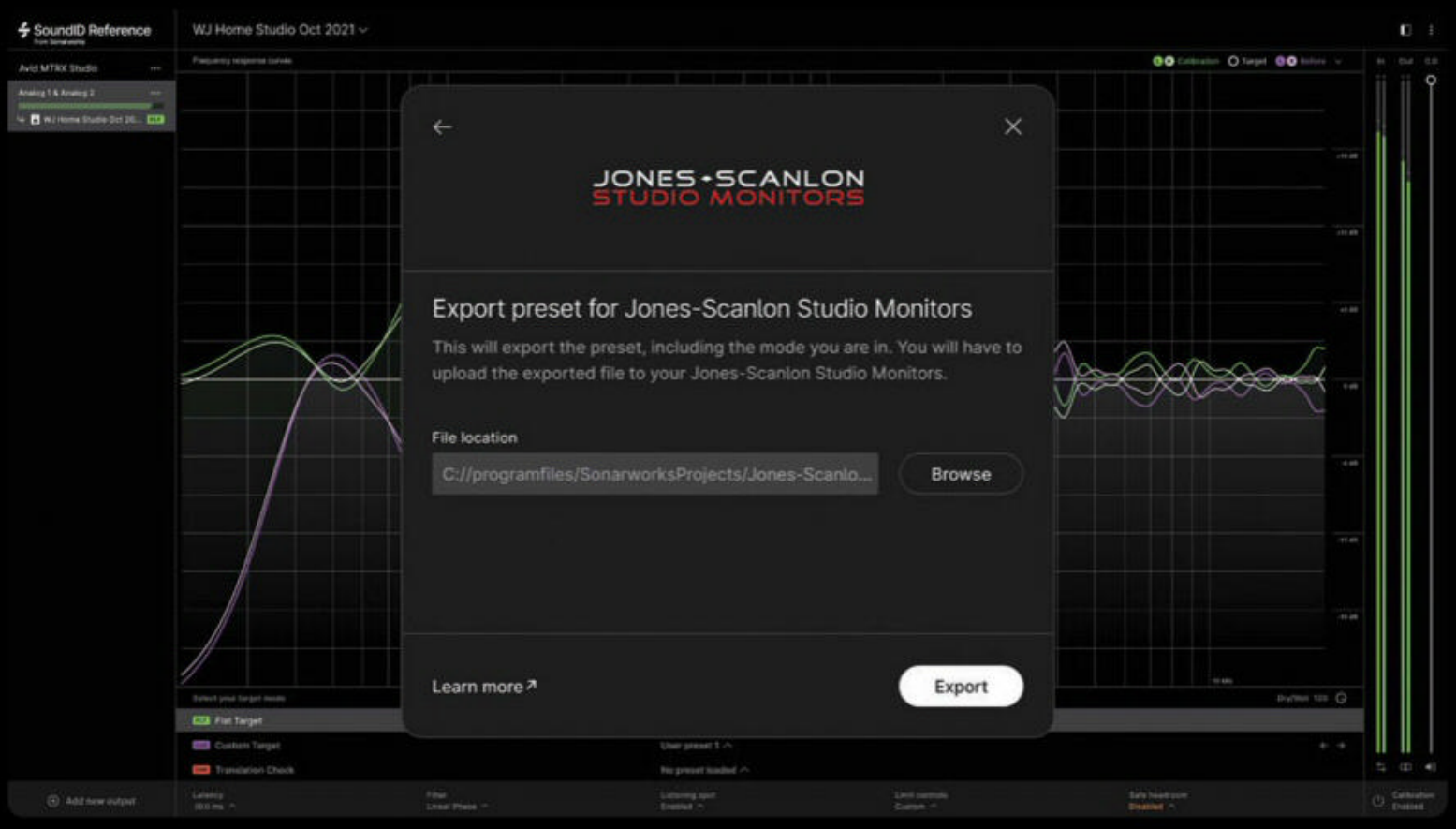

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Orchestral Tools Miroire

Sample Library

Orchestral Tools' Baroque collection shines a modern light on a historic sound.

DAVE STEWART

The latest sample library from the prolific Orchestral Tools turns the clock back to an age when people addressed each other as 'thou', said 'forsooth' instead of 'actually' and bowed and curtseyed when greeting each other before obsessively checking their phones for Queen Anne's latest TikTok post. Miroire is a collection of Baroque instruments and period-flavoured voices designed to give authenticity to historical scores while also offering a unique and versatile alternative for contemporary orchestral tracks.

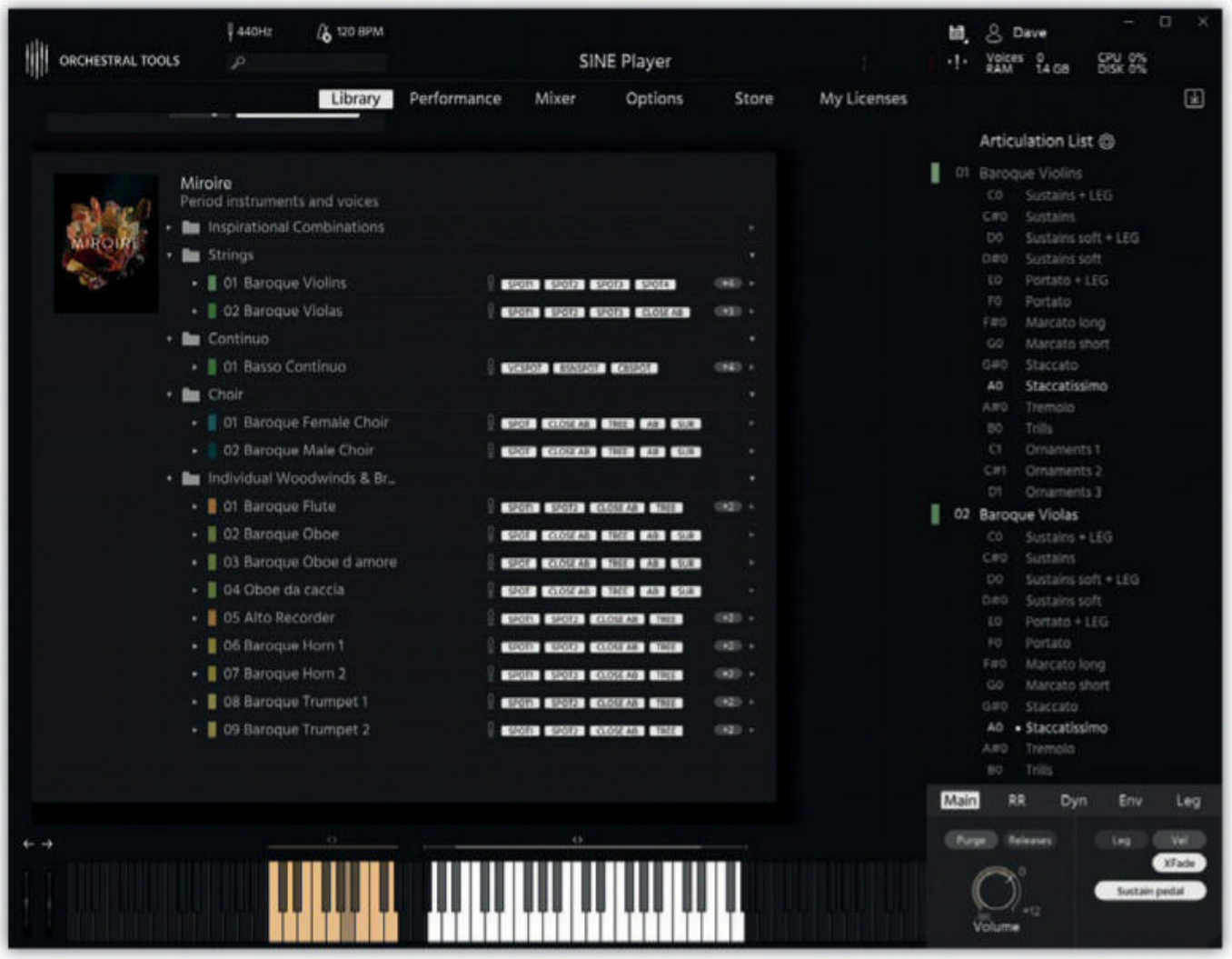
If you assumed a library of this sort would be primarily focused on recreating early music repertoire, you'd be wrong. Orchestral Tools' Sascha Knorr explains, "Miroire wasn't made to make MIDI performances of original Baroque music. Occasionally a production wants a MIDI rendition of a classical piece, and some people make classical mockups for a hobby, but in our opinion there are not enough of them to justify a commercial virtual instrument production of this scale."

Our man continues, "We're interested in the sound of those instruments and wanted to make it available for contemporary composers and productions. It's a very multi-layered orchestral sound which doesn't blend in a big warm soup like a modern orchestra, but retains a certain transparency and charmingly underdeveloped tone with its own particular expressive capabilities. But because these days our ears are quite familiar with the timbre of Baroque instruments, you can also use these instruments to quickly evoke that period atmosphere."

Overview

Miroire's instrumentation comprises four violins, three violas, a 'basso continuo' section of two cellos, double bass and bassoon, five solo woodwinds, two solo horns and two solo trumpets. These instruments are built differently from today's mass-produced items — they use few mechanical parts, trumpets and horns have no valves, woodwinds are made from softer woods and the string instruments use gut strings with no metal content. Consequently, Miroire's orchestra sounds quite different from a modern chamber ensemble, producing an overall warmer and more organic tone while retaining a powerful ensemble sound.

Also included are female and male choirs from the Vocalconsort Berlin ("bloody good singers" according to Sir Simon Rattle), who have been serving up performances of Baroque music since



Orchestral Tools Sine player makes it easy to manage instrument articulations. Keyswitches are marked in yellow and mapped samples in white — a grey line above the keyboard at either end denotes a user-programmed range extension.

2003. The instruments and voices were recorded in Berlin's Teldex Studio using Orchestral Tools' usual multiple mic positions, enabling you to seamlessly blend these samples with the company's other libraries. Miroire runs exclusively on the proprietary Sine player and is 39.3GB installed.

Baroque Woodwinds

Though we're accustomed to strings and brass hogging the limelight in today's orchestral music, for me the highlight of Miroire's Baroque sound world is its solo woodwinds, all of which span a two-octave chromatic range. The traverse flute's childlike and innocent sound is a treat. You might mistake its simple, natural tone and plain vibrato-free delivery for a recorder, but the characteristic breathy texture reveals its true identity. The instrument works well for legato melodies and piping short notes, and though its bottom few pitches sound a little fragile, the rest of the range is full-bodied, with tight, precise staccatos and staccatissimos sounding great in the high register.

Performed by the same player, the alto recorder's top notes are loud, bright and cutting. Its low notes are more reticent, but in between lies the classic sweet-toned recorder sound we all know, complete with nice breathy note attacks. The accomplished young musician plays with precision, executing uncannily accurate ornaments and trills. His woodwind performances cover all of the library's available articulations (see box below), which are implemented as consistently as possible for the other instruments and choirs.

Miroire's three solo oboes are replicas of instruments made in the 1720s by famous makers in London and Leipzig. Though examples of such period instruments are still found in museums, I'd be very surprised if they sounded better than their modern remakes — these oboes sound fabulous, and are a standout feature of the library.

My favourite is the Baroque oboe, which has a lovely plaintive and friendly tone, bright without being piercing and somewhat softer and sweeter than a modern instrument. The slightly lower-pitched Baroque oboe d'amore has a more enclosed tone, while the curved oboe da caccia ('hunting oboe') operates in the cor anglais range, producing an exotic, deep reedy sound with a hint of bassoon in the bottom notes. Beautifully and expressively played by Birgit Bahr, these oboes will enrich any arrangement.

Baroque Horns & Trumpets

Miroire's historic brass instruments are very different from their modern counterparts — they have no keys or valves and lack the complex tubing of today's instruments, so in real life can only play the natural harmonic series of their fundamental pedal note, with chromatic intervals available only in the high range. Consequently, Baroque brass players had to choose instruments in the appropriate key for the piece they were playing.

Orchestral Tools overcame these limitations by picking two differently tuned horns and trumpets and thanks to the marvels of modern science, were able to fill most of the pitch gaps by judiciously stretching adjoining samples. Two small blank key zones remain just above the instruments' low pedal notes, but this has minimal impact on their usability.

Tuned respectively in F and D, the two Baroque horns sound tremendous playing heraldic fanfares, for which you can get impressively realistic results by keyswitching between different lengths of short note. The horns also do a great »

Orchestral Tools Miroire
€478

PROS
- An authentic collection of well-played Baroque string, wind and brass instruments and ensembles.
- Female and male singers from a Baroque-inspired choir perform long notes and syllables.
- The instruments and voices blend beautifully to create an enjoyable and lively period-flavoured chamber sound.
- Programmed combinations provide instant creative inspiration.

CONS
- No percussion is included.

SUMMARY
Orchestral Tools' Miroire holds a mirror up to the past while giving a modern perspective. Its period instruments, ensembles and vocal performances create an authentic Baroque sound for historical and fantasy productions, while its unique timbres and colourful combinations offers modern composers a new take on standard orchestral textures.

Miroire's Baroque viola ensemble.

» job blasting out loud, brassy hunting-horn phrases, with a quiet dynamic available for crescendo building and post-hunt chillout music. A pair of brilliant-toned 'natural trumpets' in the keys of C and D can also dish up excellent loud fanfare-like figures and more tranquil quiet deliveries, and blend superbly with the horns in four-part brass arrangements.

Baroque Strings

While Miroire's oboes are faithful replicas of Baroque period designs, its string instruments are genuinely historic. Sarina Zickgraf's German-made viola dates from 1820, Daniel Dragonov's Gagliano violin from 50 years earlier, Lea Rahel Bader's cello was made in 1730 and the double bass played by Christoph Annaker is 300 years old! The use of gut strings, period bows with fewer hairs, the absence of vibrato and a smaller dynamic range distances these instruments from their modern counterparts. When compared with a similar-sized section in OT's Berlin Strings, the Baroque violins sound tougher and more wiry, and lack the smooth texture and metallic bow attack of the modern section.

The viola ensemble's soft sustains artic is the nearest this library gets to sounding lush, a pleasant, warm and inviting timbre which works well for chordal accompaniment and expressive legato lines. Legato patches can be switched between fingered and a more articulated bow change style; the portato legato patch adds expression to quick passing notes, and you can add accents to flowing melodic passages by applying legato transitions to the marcato articulation. A full set of short-note styles is also available for fast arpeggios and cinematic ostinatos, for which these strings' robust period timbre works extremely well.

Providing a muscular low-end foundation for the high strings is the splendid basso continuo section, comprising two cellos and a bassoon playing in unison with double bass an octave below. We've heard low octave strings before, but the addition of the low wind instrument creates a fabulous and striking big bass noise with the weight and grandeur of pipe organ pedals — one of Miroire's standout sounds. The cellos, bassoon and double bass each have their own spot mic and there's a fair degree of separation between them, so you can drastically alter the section's instrumental balance.

Baroque Choirs

Three hundred year-old singers being a bit thin on the ground, Orchestral Tools called upon members of an altogether more youthful German mixed-voice chamber choir who specialise in music of this period. The 10-strong female choir and their 12 male colleagues perform sustains, legato sustains and syllables, the latter in a choice of long, marcato and short styles.

The vowel sounds used for the sustains are 'ooh' and 'ah'. Not startlingly original, but as the two vowels are assigned to »

Articulations

This is the first library I've seen in a very long while that doesn't loop its instruments' long-note samples. In the early days of sampling that might have signalled a desire to avoid the delay and cost of creating hundreds of loops, but that's clearly not the case here. OT's explanation is that with the occasional exception of pedal notes, music of this period doesn't involve very long melody notes or chord pads, so to emulate that spirit they decided not to loop the sustains. Instead, they recorded articulations that have a certain built-in expressive dynamic movement, and are thus unsuitable for looping.

Most instrument sustains last between five and seven seconds, with the oboes' 'sustains soft' artic clocking in at eight. The melodic portato style varies in duration from 1.5 to three seconds depending on the instrument, while marcato longs are between 1.5 and 3.5 seconds. In some cases these longer notes start with a swell and feature a diminuendo in their latter half, and when you sustain a chord some notes persist a little longer than others. By contrast, the choir's long notes are sung at a constant dynamic and are looped, so you can sustain them till the cows come home.

In addition there are OT's familiar short marcato, staccato and staccatissimo styles, and all instruments except the horns and trumpets have a highly effective legato mode, as do the choirs. Grace notes, an important feature of Baroque performance, come in three flavours: the first is a double grace note called a 'lower mordent', the second is a triple grace note starting one step above, while the third is a faster rendition of the latter. All offer a choice of tone or semitone intervals. Coupled with the library's trills, these ornaments will add beautiful decoration to your courtly melodies.

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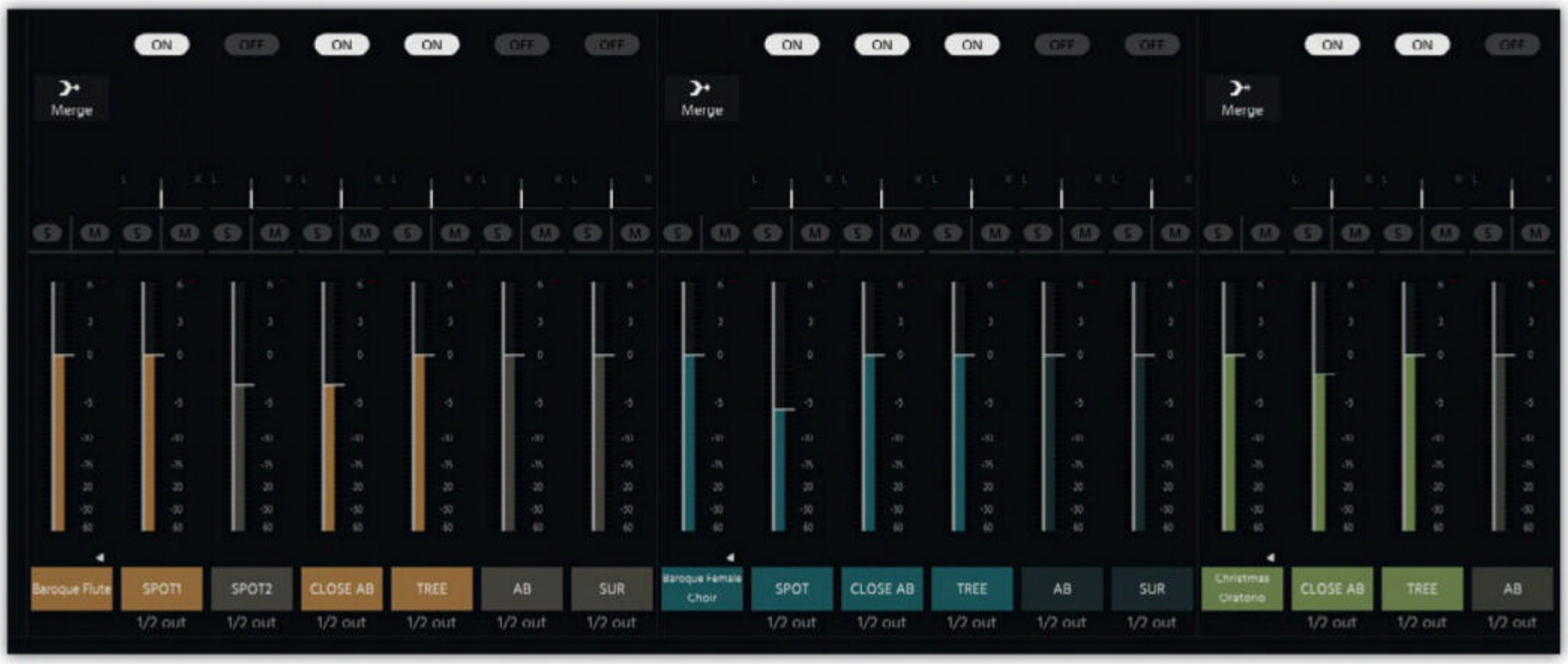

Miroire's mixer shows all loaded instruments and their mic positions. To save system resources, you can merge multi-mic set-ups into a stereo mix.

» different velocity layers within a single patch, you can create lovely evolving choral pads by using the wheel to morph between them. The long notes are well tuned and the female choir sounds great in three-part harmonies — I enjoyed the singers' steady, silky-smooth deliveries and was pleased they don't employ an overbearing operatic vibrato. I also found the legatos to be surprisingly nimble and adaptable to a wide range of tempos.

To create an impression of a Baroque chamber choir, the makers recorded a set of 19 syllables taken from German liturgical works by composers such as J S Bach. To my untutored English ear they sound like 'herr', 'komm', 'du', 'gna', 'der', 'zu', 'uns', 'sie', 'dich', 'gott', 'ku', 'mich', 'klau', 'ge', 'wo', (pronounced 'vo'), 'han', 'mensch', 'sind' (pron. 'zint') and 'tod' (pronounced 'tot'). The syllables are programmed to play in a round-robin cycle, but you can remove any you don't like or just select a single syllable. When fashioned into chords and melodies they do a very good impression of a live choir, and though the syllables are incomprehensible they're certainly a lyrical improvement on 'ooh' and 'ah'. An unexpected bonus is the marcato syllables' quiet dynamic, which has an engaging confidential quality.

Since the short syllables' initial consonants have slightly different lengths, if you put them all bang on the grid some will sound a little late. To save you the chore of manually correcting this, the makers included an optional short syllables patch in which the attacks and pre-delays of the samples have been matched up. The time-aligned samples all have a fixed pre-delay of 190 milliseconds, so to put them exactly in time you can either move your MIDI notes a bit to the left or use the MIDI pre-delay in your DAW.

Instrumentation

Strings
- Baroque violins (4)
- Baroque violas (3)
- Basso continuo section (2 cellos, double bass, bassoon)

Woodwinds
- Baroque flute
- Baroque oboe
- Baroque oboe d'amore
- Oboe da caccia
- Alto recorder

Brass
- Baroque horn 1 (in F)
- Baroque horn 2 (in D)
- Baroque trumpet 1 (in C)
- Baroque trumpet 2 (in D)

Choirs
- Baroque Female Choir (10)
- Baroque Male Choir (12)

Inspirational Combinations

Unless you're an experienced MIDI orchestrator, assembling a bunch of separate instruments into a coherent musical whole can be a daunting prospect. To help get your creative juices flowing without getting bogged down in technical details, Orchestral Tools created a set of combined patches out of the individual instruments and sections which will familiarise you with their sound and hopefully provide some inspiration.

Some of Miroire's most enjoyable material is found in this section. 'Christmas Oratorio' combines cheerful staccato oboes and trumpets with a stately low strings accompaniment, bringing glad tidings in a classic and joyous Baroque mix. In a similar vein, the celebratory woodwind-driven arpeggio-style melodies and marching basso continuo of 'Orchestra Shorts' shows off the excellent staccatissimo artic, ending with some urgent Bernard Hermann-esque ostinato chordal repetitions powered by exciting crescendo surges. Being a fan of staccato woodwinds, I had a lot of fun with this highly playable patch.

The jollity continues with 'High Ensemble Ornaments', which layers woodwind and strings grace notes to create a cartoon-like atmosphere. For more liturgical moods, the 'Passion' sustains patch combines female and male choirs with austere, serious-sounding strings, while the violins, alto recorder and female choir 'oohs' in the tritely named 'Angel Choir' sound delicate, serene and somewhat mournful. 'Wind Fanfares' and 'Brass Accents' do what it says on the tin, combining horns and trumpets in regal staccatissimo flourishes and blaring marcato accents. Lastly, there's 'Orchestra Marcato', a grandiose fusion of accented horns and strings bolstered by powerful low strings and bassoon. Reserve for your epic big screen moments.

Technical Area

I'm happy to report that the Orchestral Tools' Sine player's long-awaited

Daniel Ember plays a Baroque period horn, which lacks the valves, keys and complicated tubework of a modern instrument.

1.0.6 update is finally available. It introduces many minor improvements and fixes a number of issues, including a long-standing bug where the keyboard GUI would only trigger the first instrument in a multi-instrument setup. Disappointingly, the player still has no built-in effects and you can't tune the samples or alter the amount of pitch bend, all facilities one would expect from a high-end sample player.

The good news is that Sine makes it easy to set up your own velocity splits and/or crossfades, design your own keyswitch setups and extend instrument pitch ranges by dragging their outer notes up or down on the GUI keyboard. Another cool feature originally implemented in Orchestral Tools' 'Capsule' Kontakt script is that you can apply legato transitions to the instruments' portatos, long and short marcatos, tremolos and ornaments, smoothing out any unwanted bumps between notes.

Baroque Tuning

Nowadays we take it for granted that modern instruments are in tune with each other, so if you play a Yamaha vibraphone and your friend has a Casio digital piano, the two of you can have an enjoyable jam with no tuning clashes. This wasn't always the case — tuning in the Baroque era seemingly varied according to geographical location, and musicologists believe (for example) that pitch in South Germany was lower than in the North, while Venetian orchestras played at a higher pitch than their counterparts in Rome.

Thankfully, such differences were eventually ironed out and we now enjoy an international pitch standard of A-440, meaning the 'A' note above Middle C vibrates at a frequency of 440 cycles per second (aka 440 hertz). However, some historic variants persist: according to Orchestral Tools, today's performances of Baroque music commonly use a tuning system of A-415, and that's the tuning used for Miroire's instruments and voices. If played in literal note-for-note unison with modern instruments this would create a Portsmouth Sinfonia-style train wreck, yet the library sounds perfectly in tune with the A-440 keyboards in my music room. How so?

The saving grace is that A-415 sounds almost exactly a semitone lower than A-440. In a triumph of pragmatism over purism, Orchestral Tools mapped all of Miroire's instrument and vocal samples onto the keyboard a semitone lower than their nominal pitch, thereby uniting ancient and modern tuning systems by means of a simple MIDI transposition. It's important to understand that no sample detuning is involved — the violas' bottom string (written as C but sounding as B due to the lower Baroque tuning) is mapped to B2 on your keyboard, while the violin's bottom note (nominally a G but sounding as F#) duly appears on your keyboard's F#3. In other words, what you play is what you get. As OT's video walkthrough commentator explains, "This is just a note for the purists out there — for everyone else, don't worry: when you play a C, you'll hear a C."

Conclusion

As intimated earlier, this collection doesn't set out to be an exhaustive exploration of Baroque-era instrumentation, so if you want lutes, harpsichords, serpents, ophicleides, crumhorns, sackbuts and racketts (a personal favourite), you'll have to look elsewhere. No percussion is included, so adding an authentic beat to these period instruments would require a search for compatible timpani and tambourine samples.

Despite its relatively narrow focus, Miroire provides a satisfyingly full sound picture of a bygone musical age. Its well-chosen instruments and voices sound expressive while maintaining a simple, direct style, and the library's components blend beautifully into a united whole, providing composers with an unusual and colourful alternative. An evocative and characterful collection, and an excellent addition to Orchestral Tools' ever-expanding sound world.

£ €478.80 including VAT.
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PreSonus Sphere includes nearly every software offering PreSonus makes, including Studio One Professional and the Notion scoring suite, online storage and collaboration resources, and a vast collection of musical content, virtual instruments, and exclusive masterclasses that gets bigger every month. The Quantum 2626 is an eight-mic-input addition to the fastest family of interfaces in the world, ideal for recording drums or even a full band with lightning-fast latency performance. The new R80 V2s represent a serious upgrade to a PreSonus classic, adding acoustic tuning controls that weren't available on the original models, while keeping the same lauded Air Motion Transformer (AMT) high-frequency driver responsible for its incredible highs. Last but not least, the FaderPort V2 DAW controller offers you a full-on mixing console's worth of DAW control in a footprint smaller than your mousepad.

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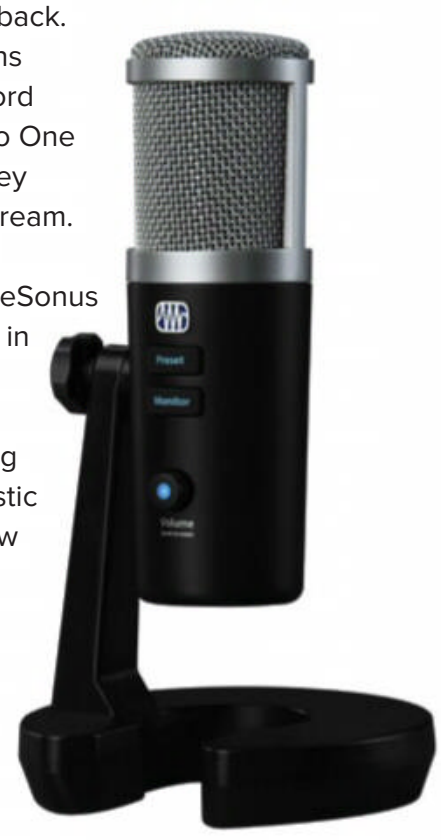
The runner-up prize, worth £295, comprises a one-year membership for PreSonus Sphere and a Revelator USB microphone. PreSonus' Revelator is an absolute must for podcasters, live-streamers, and content creators. Its advanced DSP, three selectable pickup patterns, and recallable presets bring one-click polish like compression and EQ to vocals and guitars, and its onboard four-channel software mixer lets podcasters mix in audio from their guest callers via loopback. Musicians can record to Studio One while they perform on a livestream. And of course, the aforementioned PreSonus Sphere is included in this prize as well.



To be in with a chance of winning one of these fantastic prizes, simply follow the URL shown, and answer the questions there, by Friday 4th February. Good luck!

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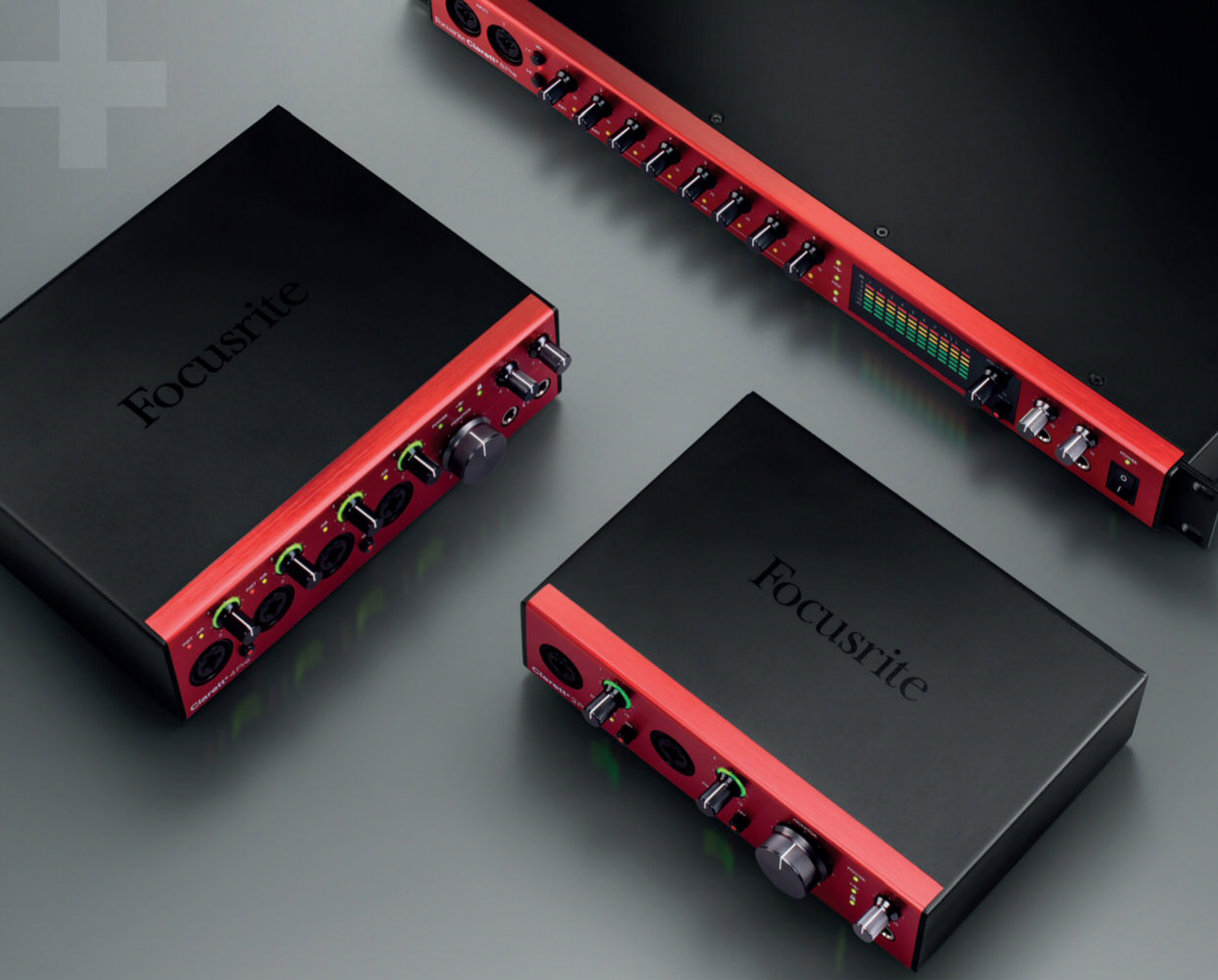

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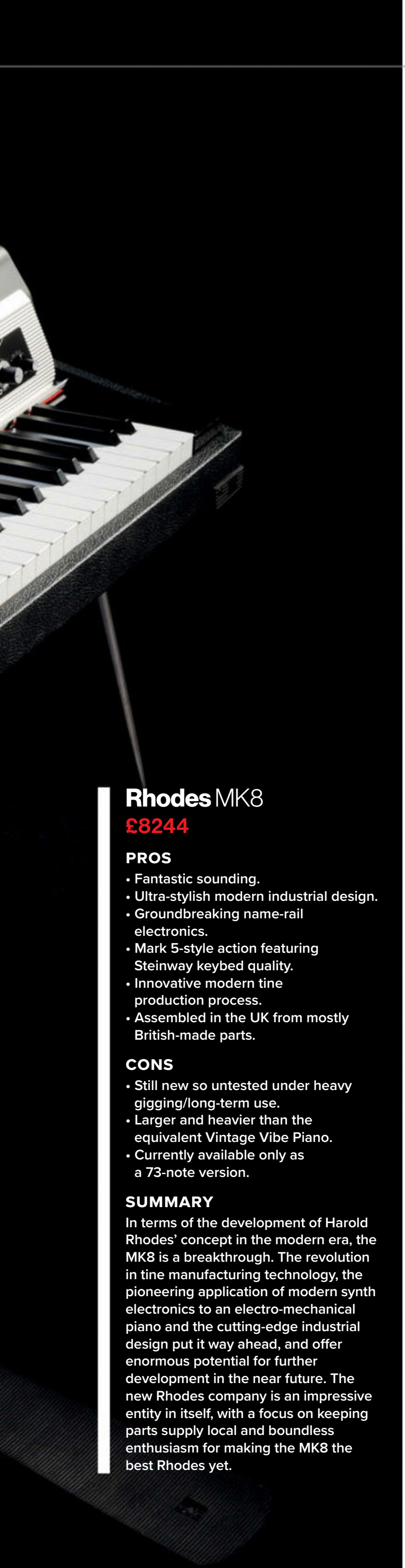
Rhodes MK8

Electric Piano

The reborn Rhodes piano is a stunning showcase for British manufacturing.



Rhodes MK8

£8244

PROS

- Fantastic sounding.
- Ultra-stylish modern industrial design.
- Groundbreaking name-rail electronics.
- Mark 5-style action featuring Steinway keybed quality.
- Innovative modern tine production process.
- Assembled in the UK from mostly British-made parts.

CONS

- Still new so untested under heavy gigging/long-term use.
- Larger and heavier than the equivalent Vintage Vibe Piano.
- Currently available only as a 73-note version.

SUMMARY

In terms of the development of Harold Rhodes' concept in the modern era, the MK8 is a breakthrough. The revolution in tine manufacturing technology, the pioneering application of modern synth electronics to an electro-mechanical piano and the cutting-edge industrial design put it way ahead, and offer enormous potential for further development in the near future. The new Rhodes company is an impressive entity in itself, with a focus on keeping parts supply local and boundless enthusiasm for making the MK8 the best Rhodes yet.

HUW REES

Initially invented to assist in therapy for injured Air Force servicemen towards the end of the war, Harold Rhodes' electric piano proved an enduring success. By the 1970s it was providing the soundtrack to an era, and went through numerous design variants or Marks before losing out to digital competition in the 1980s. Its unique soulful and expressive properties were eventually rediscovered by a new generation of musicians, to the point where a new Mark 7 was briefly introduced back in 2007. As the Rhodes renaissance gathers momentum a new Mark 8 (stylised MK8) aims to future-proof Harold's legacy. We were invited to the factory in Leeds, UK to test the new Rhodes and meet the team behind it.

Looks Are Everything

Designed by Axel Hartmann, the MK8 case looks fantastic: part minimalist industrial, part mid-century modern. Axel has chosen to evolve the classic Rhodes look, so the MK8 features the traditional wedge-shaped, Tolexed wooden case, extruded aluminium name rail and ABS lid, though the build quality is far higher than the original. Closer inspection reveals many detailed enhancements to design functionality: a cutout in the base to make space for the folding integrated stand, a removable rear logo for adjustment to the sustain pedal mechanism, and corner protectors recessed into the case. The cheek blocks, while molded in the same black plastic as the originals, have been widened, with the front left side featuring a headphone output and volume control.

Though the lid is still manufactured from vacuum-moulded ABS plastic, it's a vast step up from the original in quality terms, and will be available in multiple colours. The rear of the lid is reinforced with a metal rail which ensures stability for keyboard stacking. On our visit, lids in multiple colours already filled the factory storeroom including one in smoke grey translucent plexiglass featuring back-rail LEDs!

The name rail features a ridged, brushed-aluminium finish similar to the original Mark 1 Rhodes, though it rises higher and folds in over the front of the lid. A small updated Rhodes logo sits in the centre, while the faceplate for the electronics occupies most of the left-hand side. All connector sockets have been moved from the name rail to a plate recessed into the right-hand side of the case, so no more problems with drooping cables interfering with your funky bass lines!

During our visit we were able to see an un-Tolexed wooden case, manufactured in Sheffield from poplar plywood and super light at 6lbs. A slotted aluminium baseplate sits above the stand cutout, and the back of the case is cut down slightly to accommodate the lid's reinforcing rail. The finished base is offered in black Tolex as standard, though other colours of Tolex, as well as solid walnut, will be available as options. Overall the MK8 weighs in at 75lbs, a significant reduction over all of the original Rhodes models. At launch, it's available only in a 73-note version, though the company are considering other variants for the future.

Inside The MK8

The Rhodes team were extremely generous and open throughout our visit, allowing us to examine each component of the MK8 in detail alongside different iterations of prototypes of some parts. We were also able to get under the hood of the two prototype MK8s set up for us to test. I was even passed a screwdriver and invited to try out my own voicing!

The MK8 features active electronics designed by Cyril Lance, formerly of Moog Music, and they are by a long way the most impressive onboard electronics I've ever seen on any electric piano. Cyril prototyped the effects on his purple Rhodes, which formerly belonged to Bernie Worrell, so every parameter is calibrated to be a perfect fit for the Rhodes sound.

Sound-shaping options begin with an EQ that has a parametric mid frequency, and the option to control this either using an assignable footpedal or the onboard envelope follower, for wah and auto-wah-type effects respectively. Next to the envelope level control is a Drive knob for adding extra crunch to the sound.

To the right of the EQ is a set of tremolo controls, but this is not just an ordinary tremolo: it's more like the old Dyno Tri Stereo tremolo on steroids. In fact there are four available waveforms: square, triangle, sine and an upwards ramp which gives a striking fade-in effect. The waveforms can be assigned to both volume and envelope level, and the stereo outputs produce a series of wonderful panning effects, building on the concept of the original Rhodes Super Satellite speakers. But my favourite aspect »

All MK8 pianos will include the very impressive preamp/EQ/tremolo board. The mid band of the EQ can be controlled by envelope follower or footpedal for a wah-like effect, while the tremolo goes into the audio range for ring-mod sounds.

» of the electronics is the tremolo speed control, which at around 11 o'clock on the dial moves into audio-rate modulation. As a previous user of a Moogerfooger Ring Mod pedal on a Rhodes, I sensed a similar destructive power in this effect, and the frequency range on offer is quite similar, with the waveform options generating subtle timbral changes. Speed is set using the inner part of a dual-concentric dial; the outer part is a mixer, which ranges from a little background breakup to making the piano levitate three feet off the ground. Combined with the envelope follower, EQ and drive control, the tone-sculpting possibilities are enormous: the MK8 becomes a super-fat, fully polyphonic analogue synth controlled by the tine envelopes. Everything the old Rhodes EK10 should have been, in fact. On a gig with a MK8 my main worry would be having the self-control to use a pure Rhodes sound when this modulation potential is so easily accessible!

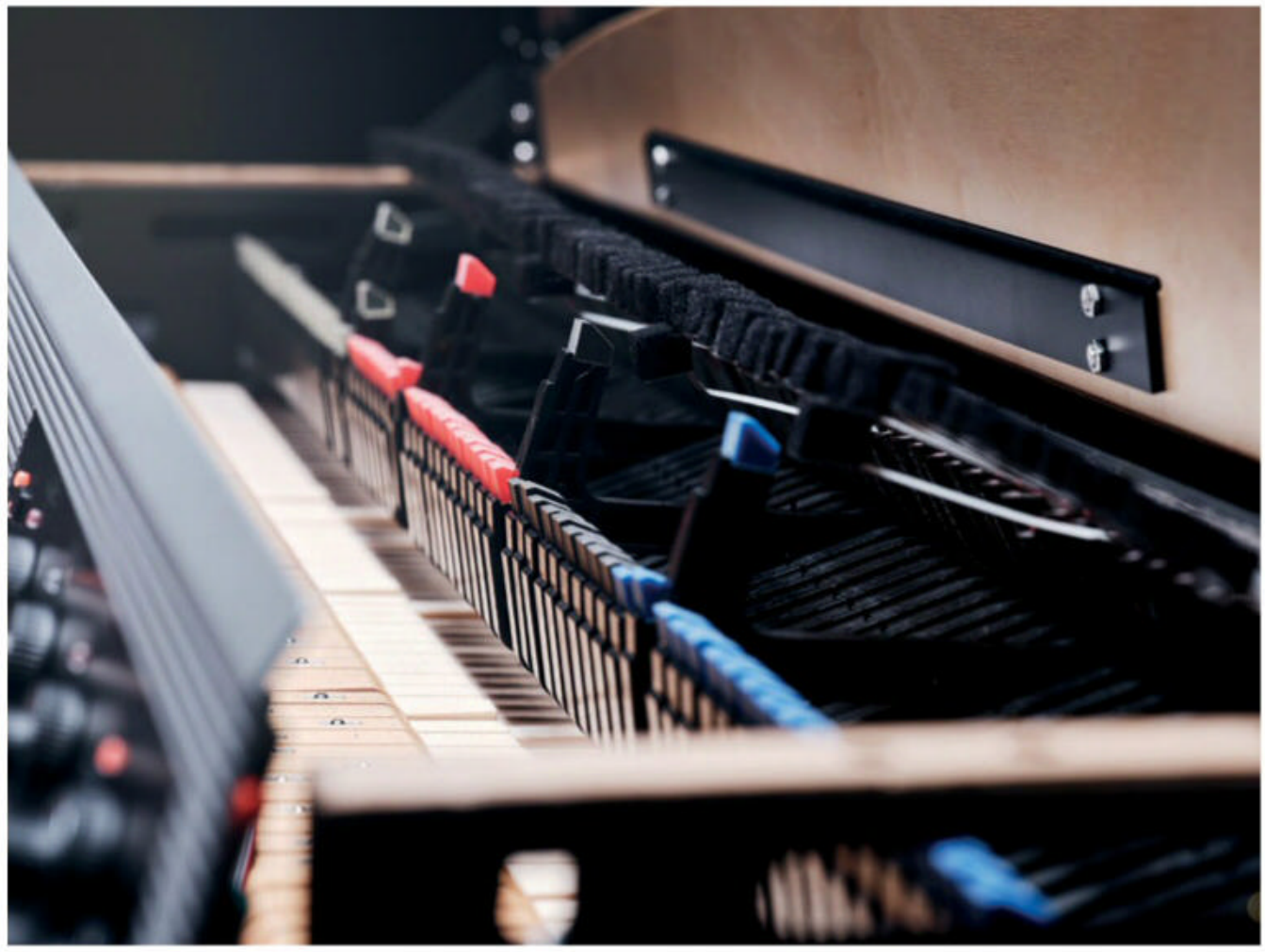
Underpinning the analogue name-rail effects is a layer of digital control. Encoder values and effects thresholds are determined by firmware, as is standard in many modern analogue synths, and this can be updated using the USB port on the right-hand side. This gives scope for enhanced and expanded functionality with each new release. Another effects module containing an all-analogue compressor, chorus, delay and phaser for the right-hand side of the name rail is available as an option. Phaser and chorus rate can be varied with control input 1, either separately or simultaneously, and delay time and delay feedback will be variable through control input 2.

Keybed & Hammers

The last production model of the original Rhodes, the Mark 5, introduced a longer keystick which allowed hammer throw to be increased for greater dynamic response compared with previous models. This design is carried over into the MK8, and implemented to a much higher standard. The keybed is one of the few elements of the MK8 manufactured outside the UK: it comes from Kluge Klaviaturen in Germany, who make keybeds for Steinway acoustic pianos. So in the MK8 we get the same premium Bavarian spruce and anti-slip keycaps as would be found in a Steinway!

The main problem with the Mark 5 design was that the extra power produced by the longer keystick proved to be too much for pivot pins on hammers in heavy traffic areas of the keyboard. There is a tendency for these pins to crack, resulting in the hammer not aligning correctly, and in some cases missing the tine altogether. This problem was solved in the Mark 7, which also adapted the Mark 5 keyboard, by increasing the diameter of the pivot pin, and the MK8 carries this solution over. Apart from being black in colour, the hammers look identical to those from the Mark 7 and are likewise from a similar high-impact ABS, self-lubricating to cope with the friction that occurs between hammer cam and key pedestal in a Rhodes. The hammer combs in the MK8 are also based on those in the Mark 7, with an enlarged hole to accommodate the larger pivots on the hammer.

Hammer tip material and hardness are important variables in Rhodes voicing. Rhodes pianos from 1970 onwards used neoprene hammer tips, which are more durable than the acoustic piano-style teardrop hammers and square felt tips found in earlier Rhodes, but nevertheless still exhibit wear if played regularly. Neoprene is still the main choice for reissue hammer-tip manufacturers today, though square felt tips as well as sets of teardrop hammers can be



This shot of the internals shows most of the key components, nearly all of which have been re-engineered in the MK8. The hammers are made of hard, self-lubricating plastic; the four raised ones show off the colour-coded hammer tips used in different hardness zones. Visible above them are the felt dampers on the aluminium damper rails. To the lower left you can see the keybed, with its long Mark 5-style keysticks.

purchased from Vintage Vibe for anyone wishing to get closer to a 1960s tone. The MK8 uses a custom-made patented polymer for greater control of Shore hardnesses and enhanced wear resistance, with the tips set at uniform height across the keyboard in four zones of hardness.

Damping

The damping system is an often-overlooked, but critical element of the Rhodes sound. The damper arms have to be stiff enough to damp the tines, but as they are actuated by a bridle strap connected directly to the hammer, must be flexible enough not to impart excessive weight to the action. To adjust damper response, the tension of a Rhodes damper arm is altered by raising or lowering the arm with the bridle strap removed. It is crucial that the damper arm material is capable of maintaining 'tension memory' once adjusted. One of the most unfortunate failures of the Mark 7 was that the damper arms were fabricated from a grade of aluminium without this property, making slack damping impossible to correct. (Having both worked on Mark 7s in the past, Dan Goldman and I agreed that even though they had some issues, the keybed was actually pretty good. Once set up with new damper arms and all bad tines replaced, a Mark 7 is capable of playing really well.)

The Mark 5 is again the template here; it improved on previous Rhodes damper arm designs by incorporating a lateral bend down the length of each arm to »

The New Rhodes Company

The main developer behind the Rhodes MK8 is Dan Goldman. Dan's passionate about Rhodes pianos, having worked with them for almost 30 years. The first part of this was spent as a performer, most notably with Morcheeba, before Dan built up his own Rhodes repair business, becoming one of the most renowned techs in Europe.

It turns out that the MK8 has partly come to fruition from a longstanding business arrangement between Joe Brandstetter, the owner of the Rhodes trademark, and Matt Pelling, now the chairman of Rhodes Music Group Ltd. In 2007, the Rhodes Mark 7 was unveiled by Joe's Rhodes Music Corporation, and it was around this time that Matt was granted rights to use the Rhodes trademark for his companies Loopmasters and Plugin Boutique. When production of the Mark 7 finished and the RMC ceased to operate, Joe's direct involvement in the manufacture of electric pianos came to an end. But as the owner of the Rhodes trademark, he was keen that any future use of the Rhodes name would only be in the development of the next generation of true electromechanical pianos. As Joe already had a productive business relationship with Matt, it seemed logical to explore the possibility that Matt could take on the manufacturing rights. After long negotiations, an agreement was reached which resulted in Joe retaining legal ownership of the Rhodes trademark, but the new UK-based manufacturing business Rhodes Music Group Ltd having exclusive 50-year rights to the Rhodes name.

Then began the process of recruiting a team to develop a new electric piano, including the vital role of a tech who could oversee the whole process, as well as making decisions about design and parts procurement. Dan was an obvious choice: as well as wide experience performing on and servicing Rhodes pianos, as the UK contact for Mark 7 service he was already highly regarded by Joe.

The actual development process began relatively recently, in 2019. Instead of resorting to purchasing parts from already established vintage keyboard parts suppliers, the intention was to re-engineer every aspect of the Rhodes concept, modifying and improving some elements of the original Rhodes, and starting from the ground up in other areas. All components of the MK8 would be directly procured by the company and developed under close supervision.

Dan has worked hard to establish close working relationships with local manufacturers, as replicating the classic Rhodes sound with modern technology has needed intensive and ongoing collaboration. Except for the keybed, all MK8 components come from UK suppliers, most based in or near Leeds. For some aspects of the MK8, numerous iterations of parts have been required, so it has been invaluable for Dan to be able to visit a supplier at short notice for a quick test of a prototype component. As much of the prototyping was initially done at small scale, it was often difficult for Dan to find a manufacturer prepared to get involved, but the suppliers who did get on board eventually became enthusiastic about the idea of reviving an iconic musical instrument! Having such tight control over the manufacturing process also allows planning for the future: Dan discussed with us the potential to develop more versions of the MK8, as well as branching out into parts supply and manufacture of other types of electromechanical keyboards.

This second effects board is an optional extra which was not ready at the time of our visit, but promises to extend the already versatile instrument still further.

» increase tension and stability at the felt without overly loading the hammer. This gives more stopping power and less chance of the damper interfering with the tine on a hard strike leading to 'spud' notes (the term used at Fender R&D for a thunking, dead sound). The damper arms are manufactured from aerospace aluminium in octave modules and, as in earlier models, are made in three sizes for bass, mid and treble. For maximum stopping power the arms have been made slightly thicker, with the bend extending as far as possible down the arm.

Choosing damper felts for the MK8 took a lot of experimentation — Dan showed us a photo of a test instrument with about 15 different types of damper felt installed. The felt that made it onto the production MK8 is soft enough not to cause a return noise on damping, as well as being woven to maintain its shape.

Tines

The core of Rhodes sound production is the tine. In order for the sound of the Rhodes to remain authentic, and each note to have consistent attack and long sustain, the design and manufacturing process is critical. From the early '70s, Rhodes tines were made using rotary swaging, a process where a steel rod is fed into a machine which hammers around its circumference. This has the effect of reducing the diameter as well as changing the grain, so that the metal is greatly strengthened. The tines are left with a taper of uniform length, in order to add more strength to the rod and guarantee a consistent strike point.

One of the most impressive parts of the MK8 project is the use of modern methods to produce a tine with the same sonic properties, but with greater durability, consistency and efficiency of manufacture. Yorkshire has a long and distinguished history of highly skilled steelworking, and Dan managed to find a firm just down the road in Leeds who were prepared to take on the project. Microscopic analysis of original tines from every generation of Rhodes production was undertaken in order to see what special characteristic imparted the classic Rhodes sound.

There is a great deal of mystique in the Rhodes community surrounding which tines have the warmest sound or greatest 'bark', so I asked Dan if any year of tine manufacture stood out as having a particularly unique quality. The answer is that all generations of swaged tine are broadly similar, though there are big variations in strength and metallurgy of the wire and block, as well as in taper dimensions and the fit of the tine to the block. From the analysis it was obvious that the traditional manufacturing method was not a precision process and that great improvements in the general assembly tolerances and machining accuracy were possible using CNC technology. The treatment process could be carefully controlled to ensure the new tines remained durable whilst retaining the correct amount of flexibility. In fact the process is so precise that it enables the tines to be manufactured in sets of 73, pre-cut to length for each note in the piano. Gone is the excess labour of feeding each tine into the swaging machine, stamping into the block, cutting to length, and quality testing. The result is a beautifully machined tine that sounds fantastic.

The immense note sustain that gave the Rhodes such an advantage over its competitors is obtained by connecting the tine to a tonebar tuned to similar pitch, in order to create an unequal tuning fork.

In this photo, the 'harp' has been tilted upwards, making visible the tines, tonebars and pickups.

Dan has stuck closely to the design of the original Rhodes tonebars, though the connecting bolts are much heavier duty than the originals and now have a metric thread. Another vital aspect of the tone generator on a Rhodes is how it is mounted. In order to maintain sustain, each tone generator assembly is mounted individually on the harp on springs, secured with two screws against neoprene grommets. Dan has opted to clone the grommets from the Mark 5, which were designed for cup washers held by a countersunk screw, rather than those from earlier Rhodes which used flat washers and a pan-head screw.

Pickups

Rhodes pianos use a separate magnetic pickup for each tine. Those made for the MK8 are broadly similar to their ancestors, but Dan and his team have made some significant improvements. As Rhodes pianos age, the point where the pickup wire is soldered to the pin becomes vulnerable to corrosion. The MK8 pickups use a new process for strengthening the wire terminations, and the pin is »

This close-up shows the tonebar (top), tines (behind the black cable) and pickups (right).

» moulded directly into the pickup, making it far more rigid than previous versions.

Rhodes pickups can be difficult to adjust precisely, due to a tendency to slip backwards or forwards as the pickup screw is tightened. In the MK8, a plastic anti-slip washer has been added to overcome this, which in my trial pickup adjustment was very effective. The metal pickup head, while maintaining a similar angle to the originals, now has a flat edge with much sharper angles than previously. The head is now held in place more firmly with lugs molded into the bobbin.

Sustain Pedal

The original Rhodes sustain pedal mechanism is a simple design. An adjustable rod pushes up a dowel to tilt the damper rail, which then rotates and pulls the dampers away from the tines. But for anyone who uses a Rhodes regularly on gigs, this system can be troublesome. As the adjustment to set pedal response has to be made every time the piano is set up, it is quite easy to push the rod in too little, leaving barely any pedal response at all, or too much, resulting in hanging notes across the keyboard. Also, the pedal has a habit of skating around the floor unless it is physically stuck down. An experienced Rhodes player will have perfected the manoeuvre of grabbing the sustain pedal mid-song with both feet and pulling it back to where it was placed at the start of the gig!

So, the MK8 sustain pedal is another aspect that has undergone a complete redesign, with Axel Hartmann again handling the cosmetics and James Cope Design in Halifax the mechanics. An adjustable metal pivot is placed at the back of the case underneath the damper rail. The sustain pedal is then connected, Wurlitzer electric-piano style, to pull down at the other side of the pivot using a cable rather than a rod. Though the pivot arm in our test Rhodes was still a 3D-printed prototype, the system was already working much better than the original rod system. A nice touch is the removable rear logo which enables access to adjust the pedal mechanism without disassembling the piano. The pedals are painted to match the lid of the piano, and they feature a rubber foot mat which folds back over the pedal for transport, neatly clipping to the 'R' logo on the pedal body.

Playing The MK8

The testing room in the Rhodes factory was set up with a variety of different Rhodes: a Mark 5, a Mark 1 'suitcase', a Mark 2 and a MIDI-equipped Mark 7 that had been extensively modified by Dan. Completing the circle of Rhodes were two MK8 prototypes, one very stylish with an olive green lid and the walnut case option, the other the main testing prototype which was finished in standard black Tolex. Sitting down at this piano, the massive step up in quality from all previous Rhodes was obvious. The standard of Tolex finish and the painting of the lid are top-class, and the keybed really does feel like it comes from a Steinway.

And how did it play? The MK8 is a beast of a Rhodes, with tons of bark and bite across the register. Getting back to the classic, warmer Fender Rhodes sound from the early '70s certainly seems to have been the priority and the tone reflects that, as does the built-in EQ: the treble dial in the front rail unit sounds calibrated more to control the high-mid frequencies in which the Rhodes bark is most prominent, so those looking for a Dyno-style treble boost will be disappointed (to be fair, no-one has used that sound since 1987). The mid EQ allows all sorts of sound-sculpting possibilities even before the envelope follower is introduced. The modulation and mixing options on the tremolo section are really incredibly musical, and I could quite happily have sat experimenting with this synth-Rhodes until late into the night. But eventually remembering that the purpose of our visit was to review the instrument and not to inflict my Jan Hammer tribute on Dan's staff, I managed to turn the tremolo dial back down to zero and get my lab coat back on. In order that we could hear right into the soul of the MK8, Dan kindly agreed to bypass the electronics and take a feed straight off the harp into one of the test amps.

In the midrange, the MK8 was still barking away like crazy. But I made sure first to test sustain pedal response, one of the most difficult things to get right on a Rhodes. Press down sustain pedal, play wide-span random chord, release chord, then release pedal very gently. The occasional hanging note while half-pedalling is very hard to eliminate, but if this is happening on multiple notes, and if certain octaves are unbalanced, then there is a problem. The MK8 performed admirably here, with clean, even note cutoffs. The stopping power of the dampers was also extremely good across the board, so that even the hardest staccato hit was stopped dead. All Dan's attention to detail with the damper arms and felts has paid off.

Escapement on a Rhodes is the distance between hammer tip and tine after the tine has been struck, but with the key still fully depressed. Due to the wide swing of Rhodes tines, especially low down, it is not really feasible to set the action to play as sensitively as that of an acoustic piano. It is possible to spend a very long time as a tech walking the line

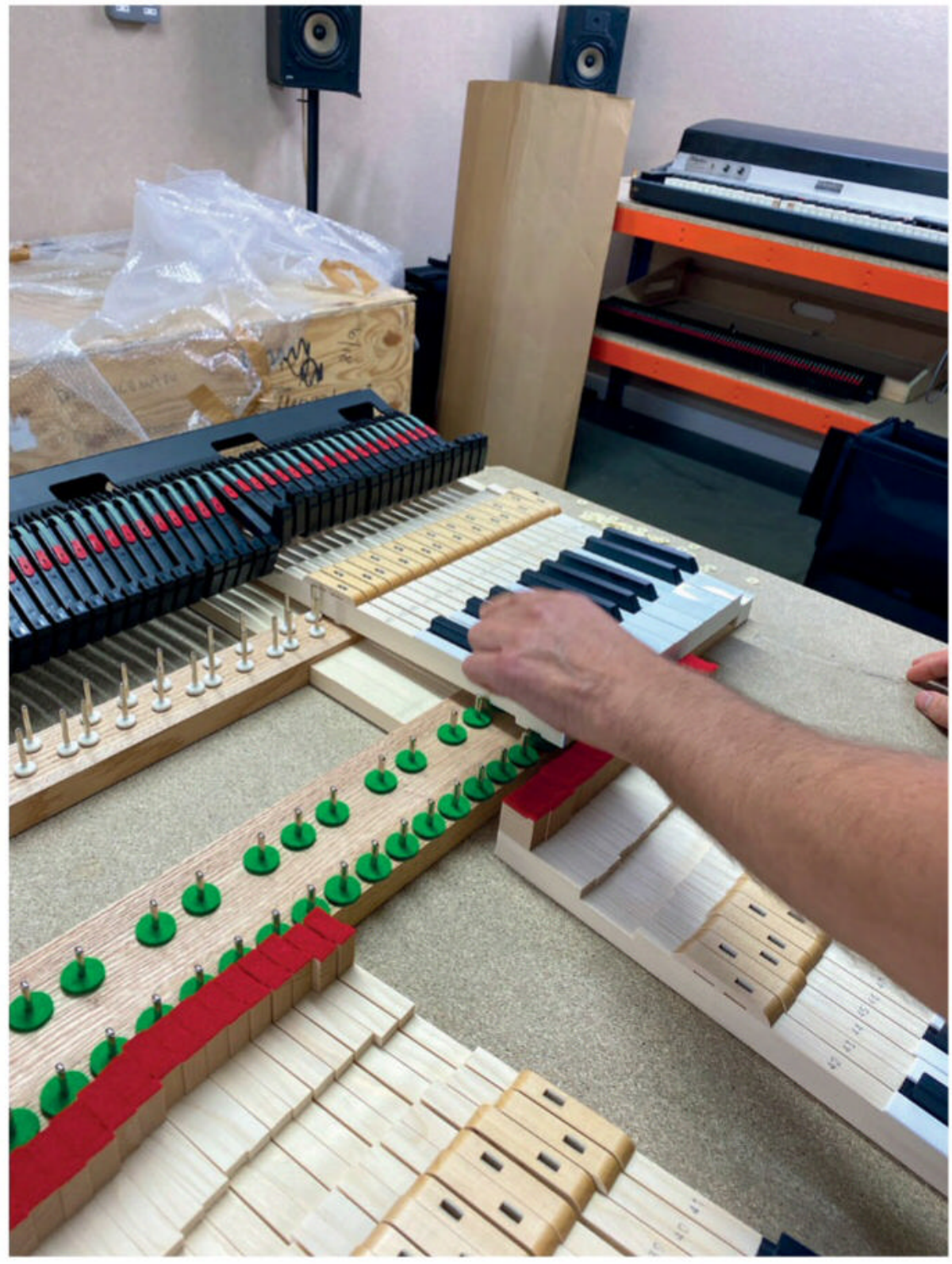
The Vintage Vibe Piano

The main competition for the MK8 comes from the Vintage Vibe Piano. This is a refined Rhodes in all but name, with the internals based on the classic Fender Rhodes models produced from around 1970 to 1975 with the half-wood, half-plastic hammers (Vintage Vibe now mold these in a single piece for extra durability). After a day servicing original Rhodes pianos at ep-service.nl I would quite often head to the upstairs studio and sit down at a Vintage Vibe Piano to reference how a well set-up Rhodes should sound and play.

Having been on the market for a number of years the Vintage Vibe Piano is now a mature product with every component tried and tested. Since it was first released in 2008, many innovations have been made to the design, particularly around weight reduction, action regulation and preamp electronics. Vintage Vibe managed to obtain the rotary swaging machine from the Torrington company on which original Rhodes tines were manufactured, so the sound of the Vintage Vibe Piano is completely authentic. And in my experience of installing Vintage Vibe tines as replacements in original Rhodes, the quality is faultless. There are many different options available; 64 or 73-note keyboard ranges, beautifully finished glass fibre sparkle-top lids in a variety of colours and a suitcase-style amplifier, as well as many different custom finishes. The exterior owes a lot to the design of the Wurlitzer 200-series pianos, though the footprint has been enlarged to accommodate a Rhodes action. While lacking the contemporary style of the MK8, the approach to weight reduction in the Vintage Vibe Piano has been ruthless, so even though it has a wooden keybed, the equivalent 73-note piano weighs in at just 60lbs. Vintage Vibe Pianos can be a little hard to come by in the UK as there is currently no established sales outlet, but it is possible to import them from the USA, and the team in the factory in New Jersey are very cool to speak with over the phone.

between a sensitive action on one hand and 'spud' notes on the other. Escapement is a matter of taste, however, and there are plenty of players who like it set with a bit in reserve. This is how Dan had set the test MK8 up, so it was a fraction high for me, but with the luxurious feel of that keybed under my hands it still felt amazing. Later I was shown a MK8 in the workshop with a lower escapement setting, so there is definitely scope for adjusting touch sensitivity to taste.

In terms of the basics of voicing, there are different schools of thought about how close the pickup head should sit to the tine and how high the tine should rest in relation to the pickup head, as well as how these two adjustments should vary across the keyboard range. Tine height regulates the ratio of fundamental and harmonics in a note. A setting too high will result in a dull, muddy sound, while a setting too low will sound nasal as excessive second harmonic is introduced. Even further and the note will jump an octave! The distance the tine is set from the pickup regulates volume and dynamic, with a closer setting giving more of both, but an overly aggressive adjustment in a search for extra bite can result in the tine striking the pickup head when hit hard. The MK8 was voiced very close, with a really nice harmonic mix, and in my brief experiment with pickup placement and tine voicing, I found the changes in timbre and volume vary pretty much as they do on an original Rhodes. Tine sustain was mostly consistent across the board: just a few tonebar clips, which increase sustain on treble tone generators, had been needed in the top octave.



The MK8 is assembled by hand in Rhodes Music's Leeds workshop.

Selling Points

So if you are in the market for a top-quality electro-mechanical piano, does the MK8 represent a good buy? It carries a high price tag, but compared to buying an original Rhodes, it has a number of advantages. First, the last Rhodes came out of the factory in 1985, so by now, most examples are in need of a major overhaul which can cost a considerable sum. With a MK8 all the components are brand new, and many, such as the keybed, electronics and case, are of far superior quality to the original. And you get basically a Mark 5 action — try finding one of those for a reasonable price in anything like playable condition! Second, customer service. Dan is very keen to build a good relationship with MK8 owners, as well as the wider Rhodes community, so there will be the opportunity to talk through the initial setup of your MK8 prior to purchase as well as plenty of tech support afterwards.

Third, the name rail electronics. To match these with another piano, you would need to budget for a few Moogerfoogers, and then still be lacking the firmware updates and the directness of the name-rail controls. Fourth, price differential. Even though the MK8 is expensive, original Rhodes have risen so much in price over the last couple of years that the difference is nowhere near what it would have been three or four years ago. Fifth, options to customise. You will be able to specify lid/pedal colour, Tolex, case material, and whether the name rail should be black or silver. Then throw in the ability to fill up the right-hand side of the name rail with effects, and maybe add to your MK8 with future innovations from the factory. Sixth, convenience. The clever integrated stand and the weight reduction, the relocation of the audio outputs and the new sustain pedal all make the MK8 very easy to live with. Seventh, and last but not least, supporting local industry and innovation. The new Rhodes company has the potential to change how music manufacturing works in the UK, as well as offering a template business model for genuine reproduction of any sought-after vintage musical instrument.

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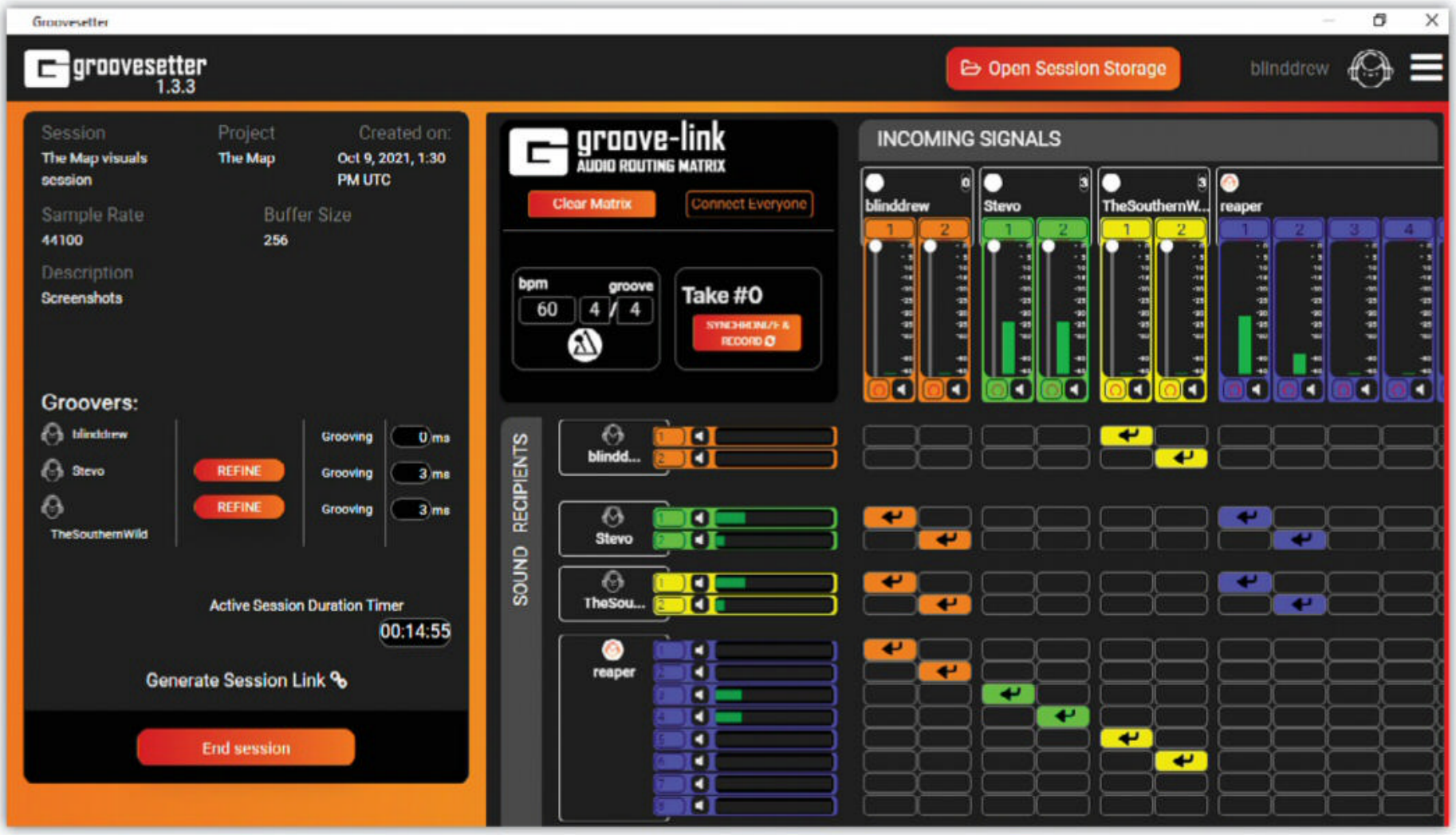


The Routing Matrix Window.

Groovesetter

Remote Collaboration Software

Groovesetter's developers claim that it offers low-latency, multitrack collaboration between multiple people in different locations. Does it live up to its promise?

Groovesetter

PROS

- Great potential as a multi-contributor, multitrack, simultaneous recording platform.
- Free version gives access to core features.
- Versatile audio routing system for all participants.

CONS

- Could be easier to set up, particularly on Windows.
- Peer-to-peer approach means cumulative latency makes real-time jamming impractical.
- Doesn't feel like the finished product just yet.

SUMMARY

This clever online collaboration software is brimming with potential, even if it's falling slightly short of its stated aims in this early release.

DREW STEPHENSON

Across the world, the Covid-19 pandemic has accelerated the rise of working from home or another remote location. It would be easy to assume that the creative industries would be more resistant to this trend, after all, there's a world of difference between sitting in a studio co-writing a song and dialling into the latest analysis of a company annual report. But a scan through a few Inside Track articles would show how the recording and production side of the music business has been working remotely for quite a while, albeit in a very back and forth way. Files, tracks and whole projects are regularly passed across countries or continents, between mixers, producers and artists to be updated, re-uploaded, and shared again.

If all of that sounds like a process that might sap the creative juices, well, you're probably right. So as we were all thrown into the world of lockdowns and travel restrictions it was no surprise to see tools like Jamulus (see our guide in *SOS* Sept 2020: https://sosm.ag/usingjamulus) and Jamkazam pop up to facilitate an online jamming/practice environment. Back in May 2020 we also wrote about running a remote recording session (Remote Music Production: https://sosm.ag/remotemusicproduction) and how you could combine a few existing tools to coach a great performance from an online collaborator.

The next logical step that we're seeing is platforms start to integrate two-way interaction with your DAW and your remote performers. And this brings us to the subject of this review: Groovesetter.

Groove Machine

Groovesetter promises the ability to mix, record and integrate multichannel, uncompressed audio from multiple remote locations, with no added latency. That's right, we're going beyond just remote stereo inputs and a two-track output to multichannel inputs and recording direct to your DAW. This means you can create a guide track in your DAW

and have multiple remote participants playing along at the same time whilst recording their individual feeds directly to separate tracks in the same DAW project. It also means you don't have to lock the singer in the booth — you can keep them in a different city entirely.

But moving beyond 'Zoom but for music' adds a degree of complexity, to both system requirements and setup and installation. For Windows users in particular, it's a multistep process, with a few potential pitfalls along the way. Having said that, during the course of this review the developers have taken on board my feedback: a new version corrected some of the issues I encountered, and further enhancements, addressing a few more hitches described below, are in the pipeline too.

I'm going to make no bones about this; setting up was, for me, a complete pain in the posterior! For Mac users the process should be a little simpler but, as a Windows user, this product is still a long way from being plug-and-play. Whilst I'm now confident that I could walk anyone through the setup process in around half an hour, it took two sessions with the developers for me to reach this point, and it's not something to be approached without a fairly experienced user at both ends and the addition of a screen-sharing/meeting tool like Zoom.

Before installing Groovesetter itself you have to install the (free) Jack Audio API, but not any version — you need the version from the Groovesetter site — and you'll also need to remember to change the install path (I've suggested that a more obvious prompt be added to the instructions at this point).

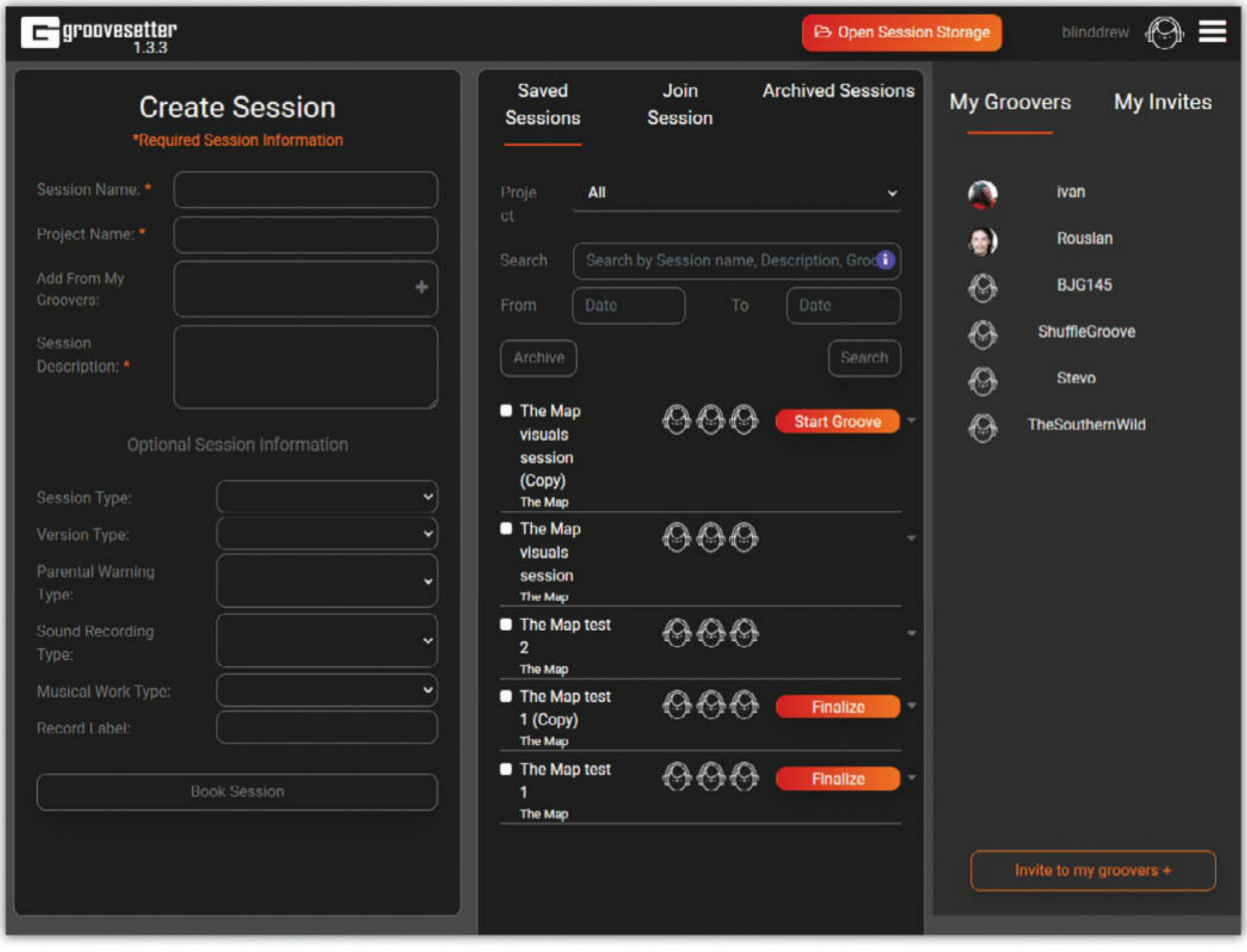
Once you've installed Jack and Groovesetter, to create or join a remote session you'll need to configure your interface in the settings for that session. Groovesetter recommend using an ASIO device as your interface in Windows, but there's still a bit of work needed on the app to automatically find and display the relevant drivers in the dropdown list. There is a manual process for this, but at present it's neither intuitive nor covered in the support tutorials (see box).

So far then, the process hasn't reached the one-app-to-rule-them-all promised land. Fortunately it does get better from here...

Into The Groove

Once you've got your coconspirators up and running, you can add them as friends, or Groovers as they're called here (strangely, this doesn't allow you to see all the extra detail that they may have added to their profile), and get on with setting up your session.

Creating a session, like completing your profile, gives you another hint that some real thought has gone into a professional's requirements. As well as fields like session name and project name (all sessions with the same project »



The Session window allows you to specify various details about the project, as well as to search for and join existing sessions.

The author, in an online session with his bandmates.

» name are automatically filed in the same folder), there are additional fields covering what kind of session it is, what type of output you're recording, parental warnings, the type of composition, record label details and there's a free-format box for general session details. It's also here that you can add your Groovers so that they'll receive an automatic invitation in their session list. You can copy and edit previous sessions – this greatly speeds up the process — which is a good thing, as you're likely to do this a few times as you refine your settings.

When you're ready to go, you start your groove from the sessions panel (or join the session if you're not the host). This opens the session settings window where you select input and output device, sample rate, project buffer size, input and output channels (up to the limit of your subscription plan and interface), and virtual I/O (up to your subscription limit) — these are your DAW inputs and outputs. The session host determines the sample rate and word length and everyone joining is automatically set to that rate irrespective of how they might have their own interfaces configured, which does cut out another potential argument...

The session window is split in two, with the left-hand window showing the basic details and Groovers, and the right-hand panel displaying the routing matrix — I'll come back to that in a moment.

The left frame will show you who's connected into the session, along with the latency (in ms) of their connection to the host, which will be driven largely by geography. In my tests, other computers on the same local network reported latency of 2-3 ms, a connection between York and Manchester (about 70 miles) around 14ms, and a connection from York to Norwich (170 miles) registered 18ms. Along with these latency reports there is the option to refine each connection by reducing the jitter buffer. This starts at a generous 74 samples but, in conjunction with the session buffer, can be brought down to as few as eight samples before the audio quality is affected.

I need to point out that "In conjunction" is doing some heavy lifting in that sentence. Although the jitter buffer can »

Video Tutorials

There's no manual (boo!) so instructions are provided by 16 short YouTube videos. I'm not convinced that the order of the playlist is the most logical, but they do give a pretty comprehensive guide to using the tool. They cover areas such as optimising latency, routing audio between DAWs and collaborators, joining and creating recording sessions, and there's a chapter on setting up Aggregate Devices for Mac users. Once you've got over the robotic script reader that does the voice over, the videos are actually a good example of how to do this well. The screen captures are clear and run at a good pace that balances being slow enough to follow what's happening on the screen without being so slow that any reminder visits become tortuously prolonged. It's a shame there aren't a couple more that cover troubleshooting the setup process.

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» be easily refined within a session (with only a short disconnection of the other Groover), changing the session buffer means ending the session, creating a new one (I told you that copy function was useful), and starting again with a new buffer size. You might need to do this two or three times, balancing the jitter buffer each time, to get to the point of minimum latency without compromising sound quality.

Once you're all connected and 'minimally latenced' (which I appreciate may not be good grammar) you move on to the routing matrix in main window. As you'd expect, this defines what audio signals are routed where with a simple pin matrix: inputs along the top, outputs along the side. As a session host, you see the ins and outs of everyone in the session, but as an attendee you see only your interactions with the session host — a factor of the peer-to-peer nature of the system.

The routing outputs default to a left-right pair, with the inputs being whatever is plugged into your audio interface. A 'Connect everyone' button speeds up the basic process. Deselecting this to customise the routing wipes the matrix, though, so in reality it isn't as useful as it would be if it left these default connections in place to be edited.

It's a simple process but as this is configurable by both the host and the attendees it actually gives a very flexible setup. For example, the host could route a stereo DAW track to L+R as usual, along with a dual-mono track from contributor A; contributor B could then choose to route the DAW track to their left ear only and contributor A's audio to their right ear. Or they could take a single mono track sent by the host to their left channel and route it to both ears. Inputs can be renamed and colour-coded for ease of navigation and consistency with any conventions you might use in your studio.

Once you've got your buffer down, your jitter optimised, and your routing configured, you're ready to go. It's worth reminding ourselves here that

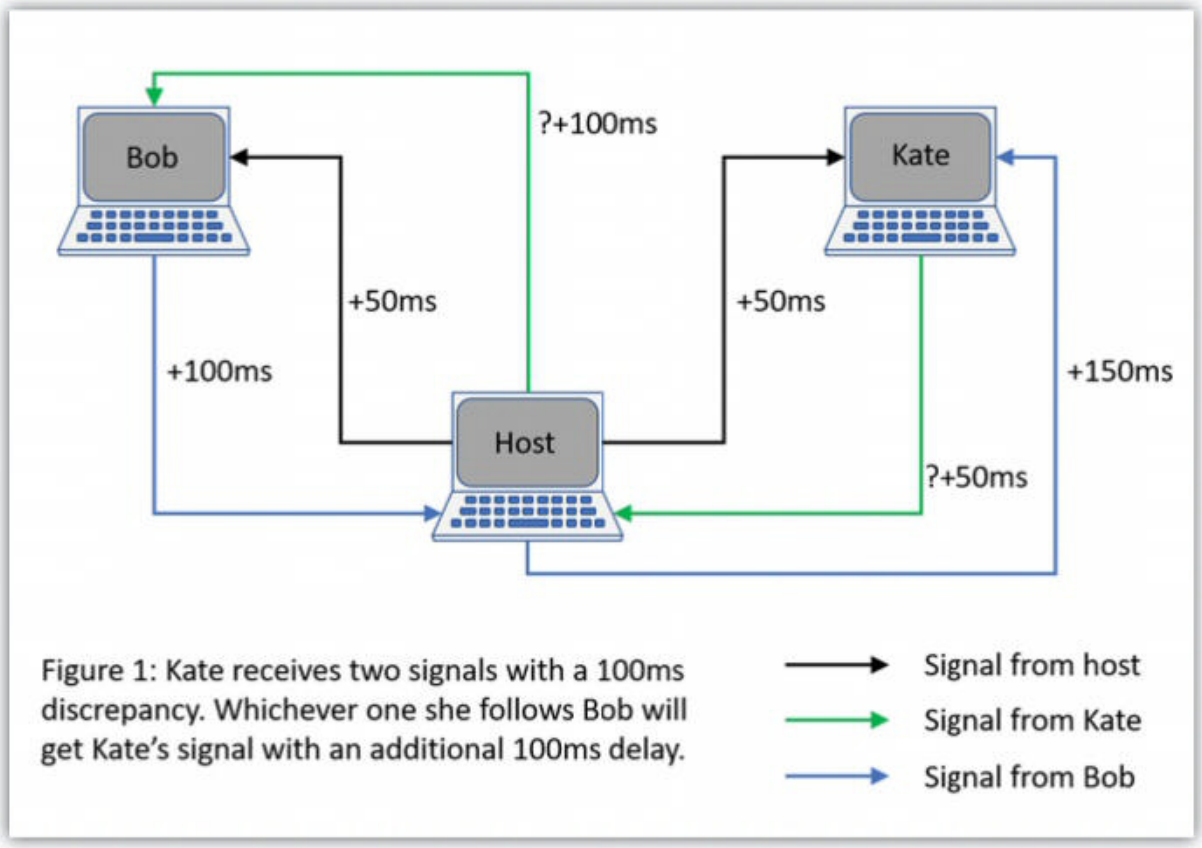

Figure 1: Kate receives two signals with a 100ms discrepancy. Whichever one she follows Bob will get Kate's signal with an additional 100ms delay.

Groovesetter is a recording tool as well as a connection tool. Each channel coming into the host can be recorded and the session window has a built-in visual metronome. Hitting record means every primed channel is saved to your session storage as a separate, uncompressed WAV file (up to 32-bit, 192kHz) in a logical folder setup (see Serious Stuff box).

You can also route any inputs back into your DAW to whatever channels you have set up and armed for recording. I mention the 'armed for recording' bit here because the only way I could get Reaper or Cakewalk to show up in the routing matrix was to have a track permanently armed for recording. I got into the habit of just tagging an unconnected track onto the end of the project and leaving it primed the whole time.

Groove Is In The Heart

Promises are easy to make but in the world of low latency notoriously hard to keep. So once you've actually hit the red button how does it work?

Unlike Zoom, for example, where everyone connects to a central hub, Groovesetter works on a peer-to-peer basis, with remote contributors connecting directly to the session host. This approach has pros and cons. The main pro is that it allows separate, optimised connections to each contributor. So if I'm hosting a session with two people, let's call them Bob and Kate, Bob's connection doesn't get slowed down if Kate's isn't as slick. And Kate won't be affected if someone accidentally unplugs Bob's router.

The big con, however, is that latency between Bob and Kate is the sum of the connections between Bob and me plus Kate and me. And that brings us to the crux of this review. In practice, despite extensive playing around (sorry, configuration) of our various buffers we weren't able to get the round-trip latency between host and contributor down to below 100ms without detriment to the sound quality. On one session we pushed it down to about 80ms but the recordings were peppered with pops and crackles. This was consistent over a series of tests with a set of fairly typical UK connection speeds of around 35Mbps down and 8Mbps up. Interestingly, one user switching between wired and Wi-Fi connection didn't appear to make any measurable difference to overall latency.

Of course, this being a peer-to-peer system meant that the latency between peers was then at least 150ms. That's too much for any kind of multi-participant jamming activity; there's no way Kate can react to Bob's part if she's hearing it 100ms after the master track or host performance. Should Bob then attempt to listen to what Kate's playing, with an additional 100ms delay, he's going to be similarly challenged as to who he's following. It's confusing enough just writing this paragraph but **figure 1** should help clarify things.

Jam Tomorrow?

So if we're not (yet) able to jam effectively, what is this software for? Well, firstly let's not underestimate the value of being able

ALTERNATIVES

In the remote jamming arena there are **Jamulus**, **Jamkazam** and **Ninjam**, and Wikipedia lists about a dozen other options as well. In the recording world, **Sessionwire** is probably the closest competitor but takes a slightly different approach in trying to automate/simplify the normal back-and-forth process.

to record multiple parts remotely from multiple locations. If Bob and Kate don't need to hear each other to play, then their latency to each other doesn't matter. And if they're playing along to a pre-recorded piece that I'm sending from my DAW, then their latency to me doesn't matter either. I'll need to do a time correction on their parts after they've recorded but that's simple enough, especially if we've taken a couple of minutes at the beginning of the session to sort out and measure the latency.

Secondly, I think this is where Covid-19 has thrown us yet another curve ball. I started this test thinking of Groovesetter as being a lockdown solution, but actually its greatest value might be in tackling the pre-lockdown challenges of remote working. Allow me to hypothesise... What if, instead of me (as a recording engineer) sending Bob the project files for him to download, import to his DAW, record his lead vocals, and then upload to Kate (as the producer), for her to do the same — making some suggestions and requesting a re-record, and so on around the houses... What if, instead of all that, we were all on the same session, listening to the same thing at the same time? Kate could coach Bob to his best performance, whilst making mix suggestions that I could apply directly to the next take. And everyone would have a copy of the session on their computer immediately afterwards. In a world of deadlines, release dates, and multiple revisions, that kind of product could save vast amounts of time.

"But what about the latency?" I hear you ask. How can Kate hear properly what Bob's doing if she's dealing with the delay of LA to New York? Well, that's actually really easy to deal with as soon as you stop trying to make Groovesetter a jamming tool. At the beginning of the session I, the recording engineer, can take a quick measurement of the latency times between me, Kate and Bob. I can then apply delays both to my playback and to Kate's feed so that we're all hearing things in time with Bob's delivery. See **figure 2** for how that could work.

Serious Stuff

In a nod to its professional aspirations and pricing, Groovesetter does have some nice features that have the potential to add value in a commercial studio. Player profile details include professional affiliations for Performing Rights Organisations memberships and these are carried across into the session metadata. I covered the useful data captured in the session details in the main text but finalising a session allows you to add a role to each contributor, from an extensive drop-down list, and to note if they're a primary or featured artist and whether they're a content owner or there on a 'work for hire' basis. Additional credits, to allow for multiple participants in each location, can be added as required. All of this is stored as an XML file in the project directory. The number of takes is also recorded and a separate folder automatically created for each one. The file-naming convention stores the project name, the session name, the contributor, the input channel, input name, take number, date and time. Automated, comprehensive data annotation like this might not make a huge difference to a home or project studio but could be a big time-saver for a commercial operation — as well as potentially keeping the lawyers at bay in the future.

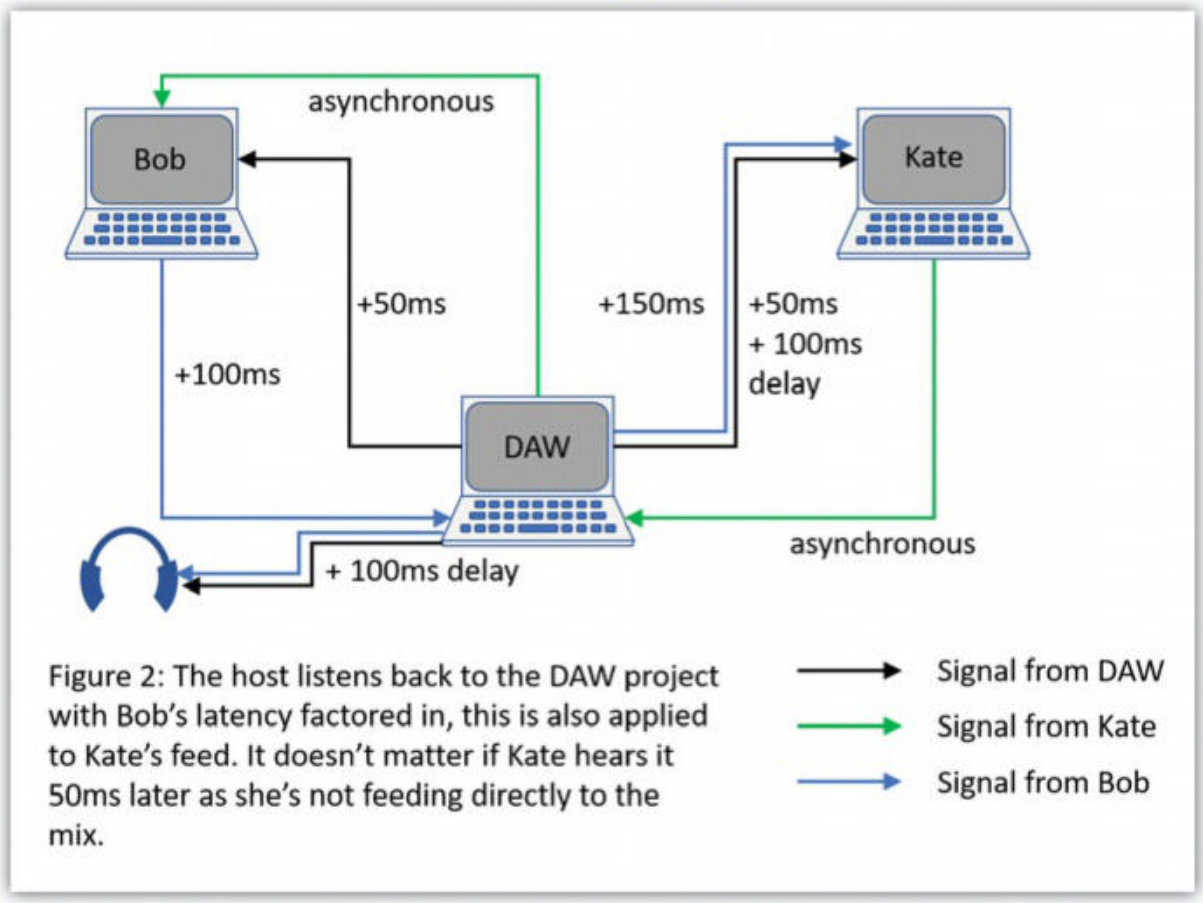

Figure 2: The host listens back to the DAW project with Bob's latency factored in, this is also applied to Kate's feed. It doesn't matter if Kate hears it 50ms later as she's not feeding directly to the mix.

The Bottom Line

When I first looked at the pricing plans I wondered who Groovesetter was aimed at. The video tutorials showed studios rather than individuals as the remote collaborators, and while I'd taken that to be artistic licence, in retrospect it's probably a better suggestion of the potential market.

I raised a number of bugs and potential improvements with the developers, and still think a lot of tidy-up work is needed if they're really to get this software up to the standard it needs to be. Yet there's clearly strong potential here. I have a final suggestion for them: as things stand, the real-world latency I experienced undermines the current advertising pitch, so stop trying to reduce the latency, focus instead on getting a smart latency-compensation tool built in (this is already in development), and pivot the marketing to support the workflow improvements that this tool could deliver to the modern professional recording world. Businesses may be changing post-pandemic, but time is still money and the potential saving here is huge.

£ Basic Plan free. Standard Plan $24.99 month. Pro Plan $45.99 a month. Custom plans also available. (Standard pricing; further discounts were available when going to press.)
W https://groovesetter.net

Not Everything Is A Nail

Sound On Sound is primarily a recording technology magazine with a strong focus on music, so that's the lens that I've cast on this review. But it's worth me pointing out that there are non-music uses for this tool too (podcasts and spoken-word recording, for example), as well as non-recording uses (music lessons/seminars), where latency is much less of an issue and the basic plan might be sufficient. I've not covered these uses in this review, but as the basic plan is free you can try it for yourself.

SSL BiG SiX

Mixer & USB Audio Interface

With the SiX, SSL shrank the large-format console. Now it's grown again!

SAM INGLIS

In the 1980s, everyone aspired to mix on a Solid State Logic console. With cutting-edge computerised automation, flexible routing, EQ and dynamics on every channel, and the famous SSL bus compressor, the E-series and G-series desks were the most powerful weapons available to the heavyweight mix engineer.

Few records are mixed on large-format consoles these days, but there's still a place for hardware mixers in the modern studio. For tasks such as rapidly setting up cue mixes, integrating outboard equipment into a DAW-based setup, communicating with musicians, managing levels and monitor control, it's hard to beat an analogue mixing desk with physical faders, buttons and knobs. The difference is that today, a console is usually an adjunct to a computer-based recording and mixing setup, rather than an alternative.

SiX Hits

SSL themselves have always moved with the times, and have proved very adept at adapting their technology to modern working practices. At one extreme, their current live sound products are highly advanced, all-digital affairs. At the other, the Origin is an all-analogue, large-format inline console that combines traditional functionality with a host of clever tweaks to make it the ideal front end for a DAW recording system. In between, their AWS and Duality are both analogue mixers and fully featured DAW control surfaces.

They've also targeted the mushrooming home and project studio markets, first with products like the X-Desk, XL-Desk and Matrix, and more recently with the most affordable SSL mixer to date: the SiX. This half rack-width console has only six inputs, but incorporates the kind of routing flexibility and master section functionality that are usually associated with much bigger desks. When you consider that the SiX features not only a pair of very high-spec input channels with preamp, EQ and compression, but also a bus compressor and a powerful monitor controller with talkback, it also represents extremely good value for money. The SiX was reviewed in *SOS* April 2019 (www.soundonsound.com/reviews/solid-state-logic-six) by

a suitably impressed Hugh Robjohns, who also confirmed its excellent technical performance using his Audio Precision test kit.

Going Large

Although it's much more versatile than such a small mixer has any right to be, the SiX naturally has its limitations, some of which no doubt have left a few potential buyers thinking "If only it did X..." Most obviously, although it's designed to work with computer-based recording setups, it doesn't have any audio interfacing capability of its own. The SiX's handling of such things as cue mixing and monitor control is exemplary, but passing signal to and from a computer means pairing it with another device. And, of course, although six inputs is plenty for many small studios, it's not sufficient for multimiking a drum kit, tracking a full band, or combining a large number of hardware synths.

For these reasons and more, SSL have now introduced a big brother to the SiX. If they'd continued their policy of deriving the name from the channel count, this new model might have ended up being called the TWeLVe, so perhaps sensibly, they've opted for BiG SiX.

The new model is BiGGeR in pretty much every conceivable way. At the most basic level, it's twice the size, spanning a full 19-inch rack width. It comes with smart and robust plastic end cheeks, but these can be replaced by an optional rackmounting kit. As on the original SiX, the I/O is divided between the front panel and a deeply recessed cavity at the top of the mixer. This is set so far back as to make access a little awkward, but means there should be no plumbing issues when the BiG SiX is rackmounted. It also does a good job of protecting the on/off switch and external PSU connector from accidentally being knocked.

The SiX employs four different connector types: female XLRs for the mic inputs, male XLRs for the main bus outputs, quarter-inch jacks for most of the other line-level I/O, and a pair of D-Subs for the insert sends and returns. On the BiG SiX, this roster is streamlined, presumably in response to user feedback. It seems I'm not the only one who dislikes DB25 connectors, as both these and the male XLRs are refreshingly absent. Instead, all line-level I/O including the main bus outputs, input sends and returns is now on balanced quarter-inch jacks. I think this is a sensible move, and while quarter-inch looms will never be as compact as D-Sub ones, they're a lot more durable in my experience.

Losing the D-Subs has meant saying goodbye to the SiX's parallel monitor output and to the analogue 'alternate inputs' for its two mic/line channels. I suspect the former is little used anyway, and the loss of the latter is massively outweighed by its replacement, which I'll come to in a moment.

Mixed Doubles

The BiG SiX doubles the SiX's basic channel count, starting with four mono mic/line channels. These have the same high-quality mic preamp, one-knob compressor, long-throw fader, fully balanced insert point and switched stereo Cue sends as are found on the SiX, but they add quite a bit of additional functionality. Each channel now has a polarity switch (hurrah!), pre/post-fade switching for the Cue sends is send-based rather than global and a fixed Mid has been added to the existing high- and low-frequency EQ bands. The channel layout is also more conventional and, to my mind, much more ergonomically friendly, with the EQ and Cue controls now occupying clearly delineated areas. The signal processing order is still fixed, with the EQ feeding into the compressor and then the insert point, but this doesn't strike me as a huge limitation.

To the right of the mic/line channels, you'll find four stereo line-level channels. Once again, these now feature pre/post switches for each Cue send, but bigger news is the addition of a three-band EQ similar to that featured on the mic input channels. The Bell options for the high and low bands are absent, and there's still no control over bandwidth or frequency, but this is nevertheless a pretty big addition in my book. It means these channels come into their own not only for combining live sources, but also for processing stems at mixdown.

Small but worthwhile enhancements over the SiX are also apparent in the BiG »

SSL BiG SiX

£2399

PROS

- Offers the same superb sound quality and technical excellence as the SiX.
- Beautifully integrated USB audio interface enables multitrack recording, stem mixing, hybrid mixing and more.
- New channel strip layout is more ergonomic and has additional features such as polarity switches and pre/post-fade buttons.
- EQ on stereo channels.
- Simplified I/O arrangements with everything on jacks or female XLRs.
- Great value for money.

CONS

- No digital I/O and thus no way to expand a BiG SiX-based system to achieve simultaneous recording of more than 12 inputs.
- Talkback setup takes some getting used to.

SUMMARY

SSL's idea of applying large-format console design principles to a miniature mixer has produced a device that is compact enough to sit on any desktop, offers uncompromising sound quality and versatility, and talks USB. What's not to like?

Each of the BiG SiX's input channels can pass either an analogue signal or a USB return from an attached computer.

» SiX's master section. The bus compressor now has a switched auto-release option, making its action even smoother on most material. New legending does away with the confusing distinction between 'Cue' and 'Foldback', although the slightly unconventional implementation of talkback takes a bit of getting used to. On most consoles, talkback is activated by a single button, with further routing options determining who gets to hear your dulcet tones. On the BiG SiX, the talkback mic is hard-wired to the two Cue busses, with separate buttons for each. These don't offer momentary operation and are rather awkwardly placed in the middle of the master section.

The master section also now boasts a second headphone socket, meaning that many users won't need to budget for a separate headphone amp. Each can be switched independently to pick up either the main monitor bus or either of the Cue busses, and there's more than enough level on offer for even the most hard-of-hearing drummer.

'Face: The Facts

The really big news, and the most significant development compared with the SiX, is revealed by a profusion of USB symbols and buttons bearing names such as From USB 1 and USB Out Post Fader. Yes, SSL have taken the hint, and the BiG SiX isn't purely an analogue mixer. It's also a USB 2 audio interface, presenting 16 inputs and outputs to a connected Mac or Windows PC. On Mac OS, it's a class-compliant device, and there is no driver software to install, nor any control panel utility to get annoyed with.

The interfacing aspect of 'mixerfaces' often feels like an afterthought, bolted on to a traditional mixer to permit basic multitrack recording and not much else. That is not an accusation you could level at the BiG SiX. SSL have thought hard about how to integrate USB connectivity in such a way as to enable multiple different working approaches — including, as I've already mentioned, hybrid or stem mixing of DAW projects. The first 12 recording inputs to the computer are drawn from the four mono and four stereo input channels. The signal is always taken post-EQ, compressor and insert point, while the aforementioned USB Out Post Fader button selects whether it should be tapped after the fader too. This is a great option to have. Tapping the signals pre-fader allows you to set up a monitor mix on the BiG SiX whilst retaining optimal gain structure to the computer, while the contrary option permits riding the fader during recording to control wildly varying levels.

Two buttons in the master section give you the option to assign USB inputs 9/10 and 11/12 separately to the two stereo Cues rather than to stereo channels 3 and 4; if you didn't need both Cues for monitor mixes, this could be used as a way of recording the talkback input, with its distinctive preset compressor. USB inputs 13/14 and 15/16 are hard-wired to pick up the BiG SiX's Bus B Master and main master output respectively, in both cases post-fader. It's a shame that these cannot be switched to pick up the four Ext In jacks instead, as that would provide a way of recording four extra line inputs at a pinch. As it is, 12 simultaneous inputs is really the practical limit for recording; conceivably you could add two more by routing only Ext In 1/2 to the master bus and recording that, but you'd lose most of the actual mixing functionality.

As I've already mentioned, the SiX's analogue 'alternate inputs' have disappeared along with its D-Sub connectors. In their place is a From USB button on each channel, which allows its signal to be taken from the USB stream rather than from the analogue input. With all these buttons engaged, USB outputs 1-4 from the computer feed the four mic/line channels and USB outs 5-12 supply the four stereo channels. The last four USB outs are routed into the BiG SiX's two stereo Ext In paths, making these the obvious choice for system sound from your Mac or Windows machine.

The BiG SiX inherits its sibling's eight-segment LED ladder meters. The topmost red LED lights when the signal

Down with D-Subs: the BiG SiX's line-level I/O is all on balanced quarter-inch jacks. The all-important USB-C socket lurks next to the PSU input, bottom left.

reaches +24dBu, and this aligns with the clipping point of the A-D converter, meaning that if you keep input signals in the green and amber zone you're unlikely to end up with overs in your computer recordings. Likewise, a 0dBFS output from your DAW will generate a +24dBu analogue signal feeding into the relevant mixer channel. The A-D and D-A converters are the same chips used in SSL's premium live sound consoles, and both offer an A-weighted dynamic range of 117dB. THD+Noise figures are equally impressive, and sample rates of up to 96kHz are supported with no loss of functionality.

Rapid Turnaround

What this means in practice is that the BiG SiX can be repurposed from a recording 'front end' to a stem mixer almost instantly, with no repatching. The potential use cases are innumerable. For example, if you leave mics permanently connected to inputs 1-4, with outboard patched into the insert points, you then have the option of printing the input signal with EQ and compression. For mixdown processing, you could then treat these four channels as 'hardware plug-ins' in your DAW; or you could stem out your multitrack across the 16 USB outputs that address the BiG SiX, mix live through its master bus with G-series compressor and optional insert, and then re-record the output into your DAW. None of these options would require any plugging or unplugging, and if you decided to add a last-minute overdub, all you'd need to do is press a From USB button. You could even hook up a tape delay and spring reverb to the Cue sends and use them for dub mixing. It's all beautifully thought out, and triumphantly extends the SiX's ethos of being more versatile than you'd expect into the digital realm.

I can only really think of one caveat that should be borne in mind. As previously mentioned, the practical limit for simultaneous USB recording is 12 inputs, and it's a hard limit: USB interfaces typically can't be cascaded in multiples, and the BiG SiX is no exception. Moreover, if you don't use the USB interfacing and try to treat the BiG SiX as an analogue front end for another interface, you won't even get 12 recording inputs from it. There's no track bussing structure such as you'd find on a large-format console, so the only way to get individual channel outputs in the analogue domain would be to repurpose the insert sends and Cues — which, of course, means that they can't then be used for their intended purposes. This limitation would be hard to overcome without making the BiG SiX a very different and much more expensive product, but potential buyers should ask themselves whether 12 input channels is really, absolutely, definitely enough for them. In an ideal world, SSL could perhaps have built in eight further channels of emergency digital I/O addressable from ADAT optical ports, for those occasions when having a stereo room mic means you can't record the hi-hat.

Special SiX

At the most superficial level, the BiG SiX might appear comparable to much cheaper products from other manufacturers: Tascam's Model 12, for example, or Soundcraft's Signature 12 MTK. But even a little investigation will show you that that's not the case. Far from being expensive for what it offers, SSL's big-small mixer actually represents amazing value for money. Other products with apparently similar features are built to a price. Their technical performance is good compared to the budget gear of yesteryear, but you probably wouldn't want to use them on a high-stakes classical recording session; and a feature set that looks impressive on paper often turns out to be incapable of adapting to different ways of working.

What distinguishes the BiG SiX is the way that SSL's heritage of large-format console experience comes through in almost every aspect of its design. You're not just getting four reasonably acceptable mic preamps that are usable at a pinch: you're getting four state-of-the-art input channels that would probably cost thousands if you bought them in rackmounting form. You're not just getting a couple of adequate headphone amps and a volume control, but a proper master section with genuinely flexible routing and monitor control facilities. You're getting a device that operates at 'pro' signal levels. You're getting balanced insert points on both the mic channels and the master bus. You're getting true stereo Cue sends. And, of course, there's that preset but great-sounding SSL EQ and compression, all of which can be switched completely out of circuit if you don't need it. Even if you treated the BiG SiX purely as a way to get hold of four SuperAnalogue input channels, and never used the rest of its features, you'd still be getting a lot for your money. When you consider what else it offers besides, it starts to look like a real bargain.

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SIDE LINES

Using Guitar Pedals In Parallel

Parallel processing while mixing is a common technique — so why not try it with your pedals when recording?

SIMON SMALL

Back in *SOS* April 2021, I wrote a feature about mixing with guitar pedals (https://sosm.ag/mixing-with-pedals), in which I covered both how to hook them up and some reasons why you might bother to do so, as well as offering some examples. In some respects, this article follows on from that one, although it applies just as much to finessing your guitar sound when recording as it does to mixing. It's about the practicalities of applying to your guitar pedal chain some of the techniques we've all grown accustomed to when mixing in our software DAWs.

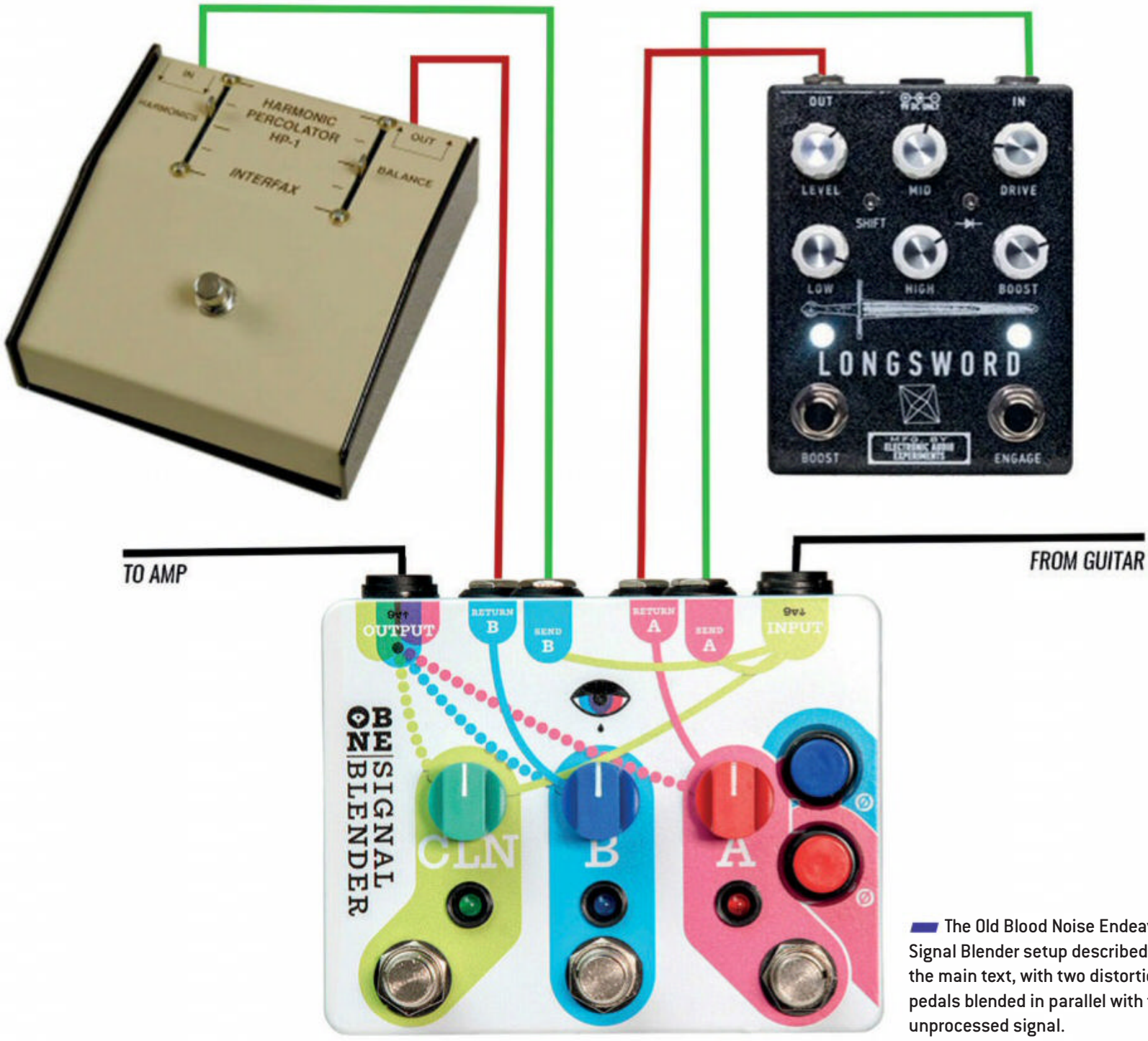
Most readers should, by now, be familiar with the idea of parallel processing while mixing. Basically, this is a technique whereby a signal path is split in two, and one path is then left untreated while the other is processed somehow, before the two are blended back together. Compressors and distortion/saturation processors are often used in this way, an approach which can raise low-level details and/or add some colour to the sound while still retaining a sense of clarity — because whatever processing is applied, the transients in the dry signal emerge unscathed.

Other than using a wet/dry mix control on individual pedals, though, guitarists tend to shy away from this tactic, presumably because it's rather easier to set up the routing in software or on a mixing desk than it is on a pedalboard. Recently, though, I've been experimenting with ways to apply parallel processing techniques to my own pedal chain when I'm recording guitars. Initially, I was searching for a certain overdrive tone and just couldn't seem to achieve it with the effects I had to hand — a couple of pedals offered some nice characteristics but, when arranged in series, they either delivered too much drive or felt unbalanced — and I really wanted to dial in aspects of each pedal's sound without them influencing each other's behaviour. So I started to look at ways that I might combine the desirable aspects of the two by arranging them in parallel and, after meeting with success, I soon started to investigate other possibilities.

Dual-Path Distortion

There are various products available for hooking up pedals in parallel and I started my experiments with an Old Blood Noise Endeavours Signal Blender. This is a simple but high-quality device with a Clean channel and two send/return loops, labelled A and B. Each of the three channels has an on/bypass footswitch and level control, and each loop has its own polarity control. So it's really easy to audition or drop out specific pedals on the fly, and to set the desired balance. It's also pretty reasonably priced (£125).

The Signal Blender was perfect for executing my initial idea of blending two overdrives: I wanted to balance the thick, low mids from my Electronic Audio Experiments Longsword pedal with the sharper top end of the Interfax Harmonic Percolator. Setting this up with the Signal Blender was quick and easy. On each loop, you simply connect the Send to the input of the pedal and the Return to its output. With that done, I could bring up the volume of both pedals, check for the best polarity relationship, and then blend to taste using the Signal Blender's pots. I really liked that once I'd set the desired balanced between the two, I could blend



The Old Blood Noise Endeavours Signal Blender setup described in the main text, with two distortion pedals blended in parallel with the unprocessed signal.

in the original clean signal to taste, to restore some of the sense of clarity which can often be lost in a heavily distorted signal such as this.

Running the two pedals in parallel gave me access to so many different tones, not least because the Longsword has an onboard EQ facility; in a parallel configuration you can shape your signal with EQ much more assertively than when the effects are connected in series. An example, in this particular implementation, is that I was able to crank the Mids control all the way up without influencing the behaviour of the Harmonic Percolator, which carried on contributing its 'top-end clang' thing, just as I wanted.

Delays

My positive experiences with distortion had me dreaming of the other possibilities opened up by running stompboxes in parallel. Using the Signal Blender once again, I soon tried setting up two analogue delay pedals: an Electronic Audio Experiments Sending and an Asheville Music Tools ADG-1. When you run delay pedals in series, they obviously have a huge influence on each other, with one applying delay to whatever comes out of the other — usually, in a guitar chain, a blend of the dry signal and the delays. This can be a great sound in its own right but in this case I wanted to create a different effect, with clear, definite delay trails at differing times, similar to the sort of thing you could achieve by using multiple aux send effects when mixing.

Running the two delay pedals in parallel allowed me to achieve this, with a quarter-note feel on the Sending pedal and a dotted-eighth note on the ADG-1. Success! The blended result was a much 'clearer' sound than when running them in series. As with the EQ on the Longsword, it's always worth thinking about what pedals can do beyond their main purpose (in this case delays) and with these pedals I found I was able to drive their preamp stages to different degrees, opening up even more tonal options. Driving the input of the Sending while keeping the ADG-1 cleaner became a favourite tactic.

Loop The Loop

While I'd enjoyed significant success with the Signal Blender, it wasn't long before I decided that it would be worth diving in deeper: I wanted access to more loops, and to have more options for interactivity between them.

With that in mind, I started to look for other gadgets that might help and this led me to the Pladask Elektrisk Matrise ($300). Conceptually, this is a fairly simple device — a matrix mixer pedal with four I/O, each with its own polarity switch, and with 16 pots on the top that you use to »

The Pladask Elektrisk Matrise, configured to run the Asheville Music Tools ADG-1 and Chase Bliss Audio Thermae delay pedals in parallel, with a little of the output of each feeding into the other.

» route your signal between the loops. But it's a hugely flexible routing system that gives you immense freedom to tweak your tone. Obviously, the four inputs mean you can run up to three pedals in parallel with the dry signal. But you also have access to cross-modulation and the ability to change signal paths on the fly, and to create feedback loops.

As a producer, having access to this kind of flexibility can be really helpful when trying to create interesting sounds during a performance. For example, whilst recording some heavy guitar riffs for a recent project, I was able to use the Matrise to feed a distortion pedal back into itself at certain points in a song. This created some heinous but wonderful, deteriorating sounds — textures that are only achievable using this method. You needn't limit yourself to distortion here. Feeding some of the signal from reverb, delay and modulation pedals back to their inputs can be a lot of fun. I also found some very strange but fun sounds when feeding the output of an Earthquaker Devices Rainbow Machine (a digital pitch-shift stompbox) back into itself.

I've found that every pedal reacts differently to having its signal fed back to its input. There's definitely a lot of fun to be had, experimenting to find the gems, but as the sound can get very loud very quickly do tread with caution!

Stereo Lab

The Matrise also has plenty to offer when you're working with stereo signals and while you could keep things simple and route a mono pedal to two outputs, it's way more fun to experiment. For instance, using the Matrise's routing matrix, you can effectively pan effects or create movement.

In one setup, I created a moving pair of delays that interacted with each other to enhance the stereo image. Loop 1 and 2 were used as outputs to different amplifiers. Loop 3 was connected to an Asheville Music Tools ADG-1 analogue delay (with its modulation bypassed) and Loop 4 was connected to a Chase Bliss Audio Thermae, another analogue delay, but with pitch-shifting applied to the repeats (in this case a fifth and an octave). I set both pedals to 100 percent wet, since the dry output was already coming out of 1 and 2 (ie. my left and right signals), and routed the ADG-1 to come out of the left channel and the Thermae from the right. However, I also routed some of the Thermae's output to the ADG-1 and vice versa. With both mono delays feeding into each other, the result was an evolving swell of pitch-shifted delays. Set the feedback to taste, and you have a really interesting effect that can be blended with the dry signal, to create a deliciously washy stereo 'ambience'.

Not every combination will sound amazing, of course, but that's the fun of this technique. It's all about experimentation and, as a producer who always endeavours to get things sounding as finished as possible at the tracking stage, I find the flexibility this approach brings really inspiring.

Bass Processing

Blending effects pedals can also be very handy while tracking bass guitar. Often, I find that putting a fuzz or distortion in series on bass can be too much and, while this effect can often be added in the mixing stage (particularly if you've captured a DI signal), I've started to blend in my fuzzes and distortions in parallel while tracking. This not only creates a slightly different sound, but it also saves me a job down the line and I find that it can influence the artist to perform differently. Compression and saturation could also be great candidates for parallel processing on bass, and while chorus and phaser pedals do typically have a wet/dry control already, this sort of routing could open up more options for them too.

BEN BAPTIE
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LEVEL UP

What Do You Get For Your Money?

The world of music technology has undergone radical democratisation over the last three decades. So is there still a reason to choose high-end over budget gear?

THE *SOS* TEAM

Today's budget equipment offers specifications that professional engineers of days gone by would have killed for. Problems such as hiss and restricted frequency response are pretty much non-issues today — so those new to recording might be wondering what advantage is to be gained by spending more than the bare minimum.

The answer is that it depends. It's certainly possible to make good-sounding tracks with highly affordable gear, but there are also good reasons why the pros choose the equipment they do. In this article, we look at five key product categories, asking what you can reasonably expect from budget options and looking at what you will gain as you move up to mid-level and professional price brackets.

Microphones

Entry Level

If you're buying a studio microphone on a budget, you have more options today than ever before, and the good news is that many of these stand up very well, quality-wise, against more expensive options. However, you'll often find deals on unbranded or obscure models that look too good to be true, and these are best avoided. If you see a multipattern valve mic with all the trimmings on sale for a couple of hundred quid, it's inevitable that corners will have been cut. Quality control will be poor or non-existent, build quality questionable, and you're unlikely to get support in the event that things go wrong.

Far preferable, and safer, to choose an entry-level model from a respected manufacturer with a reputation to protect. If you're buying your first studio mic on a limited budget, our advice would be to choose a fixed-cardioid, solid-state model. That way, you can ensure that your money is going where it really matters, and you're not compromising on the basics in order to pay for bells and whistles you perhaps won't use an awful lot.

The most obvious and versatile choice is probably a large-diaphragm capacitor mic, and there are plenty of good options that won't break the bank. These include the Audio-Technica AT2020, Tascam TM80, MXL 990, Rode NT1A, sE Electronics X1A, X1S and SE2200, Lewitt LTC240, Avantone CK6, Samson CL7A, Sontronics STC-2, Aston Origin, Lauten Audio LA-220, Sennheiser MK4, Audix CX112B and AKG C214. However, if your main applications are likely to be close-miked vocals and/or electric guitar cabinets, it's

A cardioid large-diaphragm capacitor mic is the perfect starting point for any studio, and quality needn't break the bank.

also worth bearing in mind good-quality moving-coil dynamic mics, while anyone doing exclusively acoustic instrumental recording might contemplate a stereo pair of small-diaphragm 'pencil' mics.

Mid Price

If these basic models are now so good, what advantage do you gain by spending more? In some cases, none. If, for example, you only ever use the mic for recording your own vocals, and an affordable model happens to suit your voice perfectly, there's not much reason to upgrade. However, a bigger budget does open up additional options.

You'll still find no-frills, cardioid-only designs in this mid-price bracket, which justify their price tag by using more costly capsules, transformers and other components. Good examples include the Warm Audio WA-47jr, Aston Spirit, Austrian Audio OC18, Neumann TLM102, Shure KSM32, JZ Microphones BH2, Oktava MK115, Mojave MA50 and MA201, Lewitt LTC540S and Audio-Technica AT4047SV.

A more generous budget can also get you something more versatile. This typically means energising both sides of the capsule in order to generate other polar patterns such as omni and figure-8. Multipattern counterparts to some of the mics mentioned above include the Rode NT1000, Golden Age Project FC3, AKG C314, sE Electronics 4400A, Audio-Technica AT2050, AT4050 and AT4047MP, Sontronics Orpheus, Sennheiser MK8 and Lewitt LTC441 Flex. In this sort of price bracket you also begin to encounter recreations of vintage models, such as Warm Audio's WA-414 and WA-87 R2, and Advanced Audio's CM-series mics. Valve-based mics also start to be a realistic option, with established favourites including Rode's NTK and K2, the Avantone CV-12 and Lauten Audio's LA-320.

High End

Nevertheless, there's a reason why established studio classics such as the Neumann U87Ai, Beyerdynamic MC840, Milab VIP-50 and Microtech Gefell UMT70S occupy yet another price bracket well above this 'mid-price' band. That's partly down to their status as classics, but it's also because they are still made the old way, using expensive parts and a lot of skilled labour. The same is true of many modern 'boutique' mics — but be sure to do your research before buying, because there are also cases where 'boutique' cosmetics and pricing conceal generic Chinese capsules and electronics. At the other end of the scale, meanwhile, a pro budget also makes available cutting-edge new designs like Sony's C100, Audio-Technica's AT5040, Josephson's C705 and the Manley Reference Cardioid. Finally, at both mid- and pro-level prices you'll find systems from Slate Digital, Antelope Audio and Townsend Labs that massively extend the versatility of a single microphone using sophisticated digital modelling.

As long as you only need a couple of inputs and outputs, small USB interfaces can offer excellent audio quality at a very affordable price.

Audio Interfaces

Entry Level

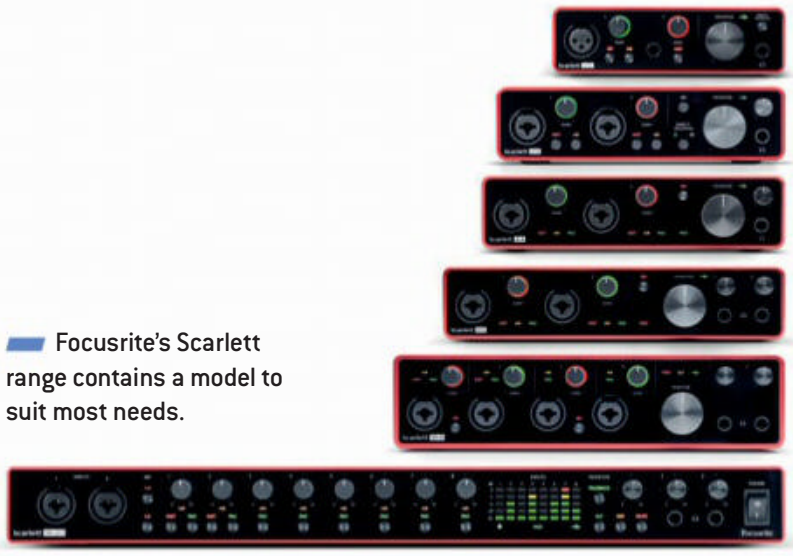
If you make music with a computer, it's rarely desirable to rely on the audio hardware built into that computer. Sound quality is often poor, especially on the input side, and you won't have the necessary connections to hook up studio mics or loudspeakers. And on Windows, most audio software uses the ASIO driver protocol, which is not supported by many built-in audio devices.

Fortunately, it's possible to make a radical improvement in this situation at a modest price. The archetypal entry-level audio interface for people who are serious about music would be a small USB audio device offering perhaps one or two mic/line inputs, a pair of line outputs for loudspeaker connection and a headphone socket. These are easy to use, can offer very impressive audio specifications, and with good driver support, will achieve low-latency performance that is more than adequate for recording with soft synths, guitar amp simulators and so on.

Once again, though, we'd caution against buying no-name or unbranded interfaces, or budget models from manufacturers who don't have a track record in this rather specialised field. There are all manner of frustrating and obscure problems that can crop up when you first dabble in computer recording, and it's definitely worth paying a little extra to access the experience and technical support that an established manufacturer will have on tap. Good entry-level options include M-Audio's M-Track Solo and Duo, Steinberg's UR12 and UR22C, Mackie Onyx Artist 1.2, the Native Instruments Komplete Audio 1 and 2, PreSonus AudioBox models and Studio 24C, Audient Evo 4 and ID4 MkII, ART's USB DualPre, Zoom's U22, Tascam's US1X2, Focusrite's Scarlett Solo, Roland's Rubix 22, SSL's 2 and 2+, and Universal Audio's new Volt 1.

Mid Price

If you only ever record one source at a time, or your music is based mainly around software instruments, an affordable USB interface may well be all you need. Once again, though, moving into what might be called the mid-price bracket broadens your options. Most obviously, it gives you access to more simultaneous inputs and outputs, which is essential if you want to record multi-miked drums, live bands and hardware synths, to integrate a modular system, or to provide headphone foldback to other musicians. With Steinberg's UR-C series, Focusrite's third-generation Scarlett range and PreSonus's Studio interfaces, the affordable interfaces mentioned above represent the entry point to ranges containing several models with »



Focusrite's Scarlett range contains a model to suit most needs.

PreSonus's Quantum 2626 offers Thunderbolt rather than USB connectivity, for lightning-fast performance.

different I/O counts. In each case, the top-of-the-line model features eight mic/line inputs, eight or 10 line outputs, ADAT-format digital expansion and more, and a broadly similar feature set can also be found on Arturia's AudioFuse Studio, Tascam's US16x08, MOTU's 8Pre USB and Zoom's UAC8.

Moving up to the mid-price region of the market also introduces numerous extra possibilities in terms of features. For example, the larger interfaces in Audient's ID range combine multichannel I/O with a console-like monitor control section; Antelope Audio's Zen Go incorporates their Synergy Core platform for running low-latency effects, processing and mic modelling; and UA's Volt 76 models feature true analogue compressors based on the classic 1176 design. Finally, a mid-price budget also means you can consider Thunderbolt interfaces, which are typically more expensive than their USB brethren for the same I/O arrangements, but may offer better low-latency performance and other practical advantages. You could also step up to an interface with even better audio specs and quality, such as Black Lion's Revolution 2x2, Apogee's Duet 3 or Focusrite's Clarett range.

High End

Feature sets become much more diverse when you get into the pro price bracket. There are relatively few models that offer more than eight built-in mic preamps, because the expectation at this level is that users will want to integrate their own preamps or mixer. So you can, for example, find interfaces such as the PreSonus Quantum 4848, MOTU 16A, UA Apollo 16x, Focusrite Red 16Line, Prism Sound ADA128, Apogee Symphony and Antelope Orion 32 that offer mainly line-level analogue I/O, along with numerous models from RME and MOTU that focus on digital expansion formats such as MADI, Dante and AVB. There are also products such as RME's Fireface UCX+ and UFX+, Apogee's Ensemble Thunderbolt, Merging's Anubis+ Music Mission, Prism Sound's Titan, Avid's Pro

Masters of all trades: a pro budget can get you an interface with enough flexibility to deal with most recording scenarios.

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Not bad for free! Most DAWs now come with capable drum instruments, such as MOTU's Model12.

Many modern drum instruments include not only great drum samples, but also a virtual player to perform them.

» Tools | Carbon and the Antelope Zen Tour Synergy Core that offer various deluxe takes on the 'Swiss Army knife' idea of an interface that can tackle any scenario.

Drum Instruments

Entry Level

Whether you're using them to provide songwriting inspiration, as a placeholder for a real drum performance yet to be recorded, or to create finished drum tracks, software drum kit instruments are a key part of music production today. For this reason, most DAW software comes with pretty capable digital tub-thumping options at no extra cost. Apple's Logic and Steinberg's Cubase have intelligent 'virtual drummer' instruments in the shape of Drummer and Groove Agent SE respectively, while PreSonus's Studio One and Ableton Live come with more universal sampler instruments that are also highly capable sources of drum sounds. MOTU's Digital Performer has the powerful Model12 drum sampler, while the Avid Complete Plugin Bundle available with Pro Tools subscriptions includes the AIR Boom instrument. There are also innumerable freeware or shareware VST plug-ins, samples and soundbanks around.

Mid Price

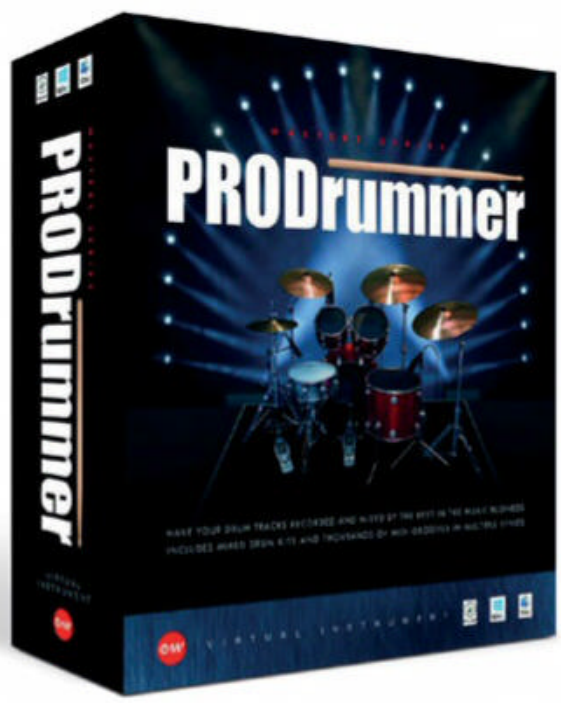
These bundled plug-ins are all very capable, but even a modest budget targeted specifically towards drum software will open up other possibilities. Depending on how you spend it, you could target flexibility of sound at the mix, a wider variety of sounds or MIDI patterns, or perhaps sounds and styles appropriate to niche genres.

For example, UJAM offer a great variety of 'virtual drummer' plug-ins which are oriented towards specific musical contexts such as funk, soul, hard rock and so on, while Native Instruments' range includes the likes of Butch Vig Drums and multiple Abbey Road Drummer libraries. Steinberg likewise offer an extensive catalogue of affordable expansion packs for Groove Agent SE — which is compatible with any VST-supporting DAW — that dive deep into the sound and feel of particular genres.

High End

Up your budget to three figures and the world of sampled drums is your oyster! Options such as Toontrack's EZ Drummer, NI's Studio Drummer and XLN Audio's Addictive Drums prioritise ease of use, letting you get a great sound fast, while engineers who demand the ultimate in control can explore the likes of Toontrack's Superior Drummer, FXpansion's BFD and Slate Digital's Steven Slate Drums 5. You'll also find instruments that have a particular USP of their own, such as Spitfire Audio's The Grange, Platinum Samples' Ocean Way Drums and EastWest's Pro Drummer. Companies such as UJAM and NI also offer collections of their mid-priced libraries at very cost-effective prices within this bracket.

Perhaps the best value of all, though, is to be had through subscription packages and bundles. PreSonus's Sphere, for example, is much more than an affordable way to own Studio One and includes a huge amount of sample-based content. Likewise, EastWest's Composer »



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Portable PA Systems

Entry Level

With many tech purchases, there's a strong argument to adhere to the old 'buy right, buy once' maxim, but as with microphones, once you start getting heavily involved with live sound you can't have too many PA speakers! The budget PA you buy when you're just starting out will always come in handy, even if you upgrade your main system, as you'll often encounter situations where you need an extra stage monitor, side fill or drum wedge.

Budget PA systems have come a long way in the last 10 years or so, with most of the big names — Mackie, Yamaha, JBL and so on — offering good-quality, lightweight active speakers with ample power for small gigs that won't break the bank. Even the most affordable active system, such as those from DV's Fame range, will be more than capable of elevating keyboards and vocals above the sound of a full drum kit.

Mid Price

Of course, your options open up as your wallet fattens. If sound quality is your biggest concern, then a little more money may net you a speaker with better (or more) drivers, or a wooden cabinet, which will tend to sound better than an equivalent plastic-box speaker. Brands like Martin Audio, PreSonus, RCF, QSC and Electro-Voice are popular choices for people looking for a step up in quality.

Around the £1000 mark, compact line-array-type systems, such as Electro-Voice's EV50M, HK Audio's Polar series, Yamaha's Stagepas 1K, the Fishman SA330x and the Mackie SRM-Flex become available. These sorts of rigs are extremely portable, and also tend to incorporate built-in mixers, which saves you having to lug a separate one around. As a bonus, they're also very resistant to feedback, which has made them popular in the pub/café open-mic-night scene. The LR Baggs Synapse portable PA speaker doesn't quite operate in the same way as a line array, but offers very similar benefits thanks to its extra-wide dispersion.

High End

The really posh stuff tends to be intended for use in larger multi-speaker arrays, though there's nothing stopping you from assembling a small, compact, high-quality rig from premium brands such as d&b audiotechnik, Meyer Sound and the like. High-end column arrays are also available, and if you want to combine top quality with the convenience of a portable line array, then brands like Fohhn and FBT are the ones to watch.

Monitors

Entry Level

The affordable end of the monitor market caters to a wide range of people, from amateur podcasters to gamers who dabble in the dark arts of the DAW. These so-called 'multimedia' speakers tend to be active (thus not requiring a separate amp), keenly priced, and compact enough to sit on a typical home office desk. They may not be the most accurate speakers around (for that you'll have to pay more), but their manufacturers understand the need for convenience and ease of use. For example, the smaller PreSonus Eris models, JBL's 1 Series, and Mackie's CR-X series of desktop speakers are available with Bluetooth connectivity, so you can stream music from your phone, tablet or laptop; while the Alesis M1Active 330 USB even incorporates an audio interface.

Many of these multimedia speakers also include a headphone output, and

Buying an affordable PA no longer means settling for bad sound!

A mid-price budget can get you excellent performance from either conventional or miniature line-array PA systems.

In the pro bracket, you can now buy small systems that share their technology with stadium-sized PAs.

Reference-quality monitoring is now more affordable than ever, with previously 'pro' features such as three-way monitoring and coaxial design available at mid-price budgets.

A four-figure monitor budget can get you powerful DSP assistance as well as excellent electro-acoustic performance.

they also tend to have front-panel volume controls, meaning you won't need to buy a separate monitor controller. If you're just getting started with music-making then a pair of affordable desktop media speakers should see you right for some time with a minimum of fuss and cost.

Mid Price

When you've got a little more to spend on monitors, however, you're expected to already have things like monitor controllers and audio interfaces. So, more expensive models may lack some of the conveniences of multimedia speakers, but they tend to be better engineered, made of more expensive materials, and be better at reproducing sound accurately.

Between, say, £400 and £1500 per pair, there's a huge choice of monitor formats available. There are some excellent traditional two-way, ported designs, like the Avantone CLA-10A, Dynaudio Lyd series, Focal Alpha Evo range and ADAM AX series, as well as coaxial models from the likes of Fluid Audio, KEF and Output. Passive radiators become an option at this sort of price range too, thanks to the Reproducer Epic 5 and Mackie HR MkII series. Even three-way monitors have started to become affordable now, thanks to models like the Kali Audio IN-5 and KRK Rokit 10-3 G4. For more info on what sort of monitor would suit you best, see our recent feature about buying studio monitors at www.soundonsound.com/sound-advice/how-choose-studio-monitors.

Value-oriented 'multimedia' speakers can make effective studio tools, and frequently offer additional features.

High End

The rarefied end of the market can broadly be split into two. On the one side, you've got entirely passive loudspeakers from manufacturers such as B&W, SP Acoustics, Amphion and ATC, designed to be paired with the power amp of your choice. Those brands and their peers focus on the electro-acoustic design of their speakers, with the cost justified mainly by their use of world-class drivers made from expensive materials.

The other side of the high-end market represents a marriage of this electro-acoustic expertise with modern digital technology. Genelec, for example, employ extensive computer modelling in the manufacture of their speakers; and their monitors also incorporate advanced room-correction DSP, as does ADAM's S-series and models from Barefoot, Kii Audio and Jones-Scanlon.

Studiologic Numa X Piano GT

Stage Piano

Studiologic bring their A-game with a seriously capable new stage piano.

DAVE GALE

On several occasions, I have found myself taking a tape measure with me to a car showroom, specifically to establish whether my 88-note keyboard will fit in the back of a prospective new car. Of course, that's only part of the 88-note journey; as any keyboard player will tell you, the more realistic the action of the keyboard, the heavier the beast becomes, and that's no joke when you're traversing from one gig to another on a weekly, or even daily basis. Feel free to speak to my sore back for confirmation.

Over the last 20 years, keyboard technology has marched on, with ever-better and realistic keyboards which don't necessarily weigh more than the UK National Debt! Take a bow Fatar, the Italian company that have kept many of the World's finest electric pianos and synths populated with top-quality keybeds. Studiologic are effectively Fatar by stealth, the major difference being that the Studiologic brand is more commonly associated with full-blown controller keyboards, synths and pianos. It's the piano angle that we're interested in today, with the Numa X Piano range, but as we'll discover, there's far more on offer than a basic stage piano.

Nice Action

The review model is a pre-production Numa X Piano GT, which is the upper-most model in the range. There are also two smaller models, which drop in size, weight and price (see the 'Little Brothers' box). This top-of-the-range model is an 88-note workhorse, weighing in at 21kg. The chassis is metal, and as solid as they come, with a full-sized keyboard action. Moreover, given the association between Fatar and Studiologic, this is the first outing of their newly designed action, which is known as the TP-400. This is an update to the previous and much-lauded TP-40 bed, found in many other high-end keyboards.

For some years, I have been a Studiologic Grand controller user, which employs the older TP-40 keybed. This became an instant favourite with me, but with a newly sharpened focus, the difference between older and new actions is relatively pronounced. The TP-400 feels less spongy to play, with a greater degree of assurance if playing softer or more sympathetically. The keys go down relatively easily, but there is enough resistance there that you have to work. As your fingers move toward the fulcrum point, there's a nice balance; keys still go down easily, but like a piano, you'll need to work a tad harder.

Part of the reason for this overall sensation is to do with the introduction of an escapement component, which counterbalances the key over the fulcrum point. It's a very realistic concept, with roots based in acoustic piano designs.

It's important to point out that with any keyboard of this kind, it far extends the mere scope of a stage piano. It's a perfect controller keyboard, and it'll perform brilliantly at home or on stage. Realistically, taking it to a stage setting means that you'll be more reliant on its core sound engine, which is our next port of call.

Engine Room

The Numa X Piano sound engine is pretty easy to grasp. In essence, it's all about

The Numa X Piano GT's back panel is generously equipped with three quarter-inch pedal inputs, four quarter-inch audio inputs, quarter-inch stereo output and headphone port, full-size MIDI I/O sockets and a USB port.

sample playback. As a starting point, 220 Factory patches are included, constructed from a vast number of sampled instruments, which at the time of review were still being added to. There are Steinways, Yamahas and even a Fazioli, alongside contemporary instruments such a models of Rhodes, Wurlitzer and electric grand. There are also upright pianos, some in a state that could be regarded as slightly out of tune, along with synths, incredibly realistic Hammond organs, string and orchestral sounds, accordions and choirs. In fact, it's probably easier to list the category of sounds which aren't included.

Sticking with the more common stage piano instrumental sounds, there is a striking contrast across models. While the German (Steinway) sounds rich and full toned, the Italian Grand (Fazioli) is very striking in the lower mids. They offer what I would call an assured depth of tone, with a lovely sense of strike, as the hammer hits the virtual string. The vibrancy of the Japanese Grand (Yamaha) cuts through the mix brilliantly, with a sharpness of upper timbre that is also exhibited in the lower note range too, as Yamaha's do! Overall note tuning is equally tempered, but there are moments of organic reproduction, such as the main Upright sample. This feels slightly 'twangy', but is characteristic of a particular flavour of upright, and perfect for more modest musical outings.

Meanwhile, the Rhodes, Wurlis and Clavs are all exemplary, with plenty of model versions to audition. There's even a great sample of the Yamaha CP-80, as well as numerous synths, string machines, pseudo-acoustic ensembles and instruments. It's very comprehensive indeed!

Sample selection and audition takes place within a patch, with category buttons guiding your path. Loading is pretty instantaneous, which is both helpful at the editing stage, and for live, where milliseconds count if switching patches mid-song. Studiologic have created a Favourites section too, so you can place your preferred patches in close reach, via the dedicated Favourites button.

Construction Time

Prior to review, Studiologic went to great lengths to point out that the Numa X Piano factory patching only soaks up half of the on-board memory. There is plenty of capacity to load future sounds, after initial purchase, through the use of their free Numa Manager editing and librarian software. This will make light work of patch organisation, as well as allowing further samples to be uploaded to the piano itself, which will be available from the SL website. There is no provision for user sample import just yet, but there are plans for plenty of other sounds to be available, drawn from pianos to synths and acoustic instruments.

Another point that Studiologic were keen to highlight is the economic way that the samples are used. There is some very clever technology in the Numa X OS which takes full advantage of sample compression, meaning that even the most comprehensive of instrument samples is relatively lean. This is excellent news, as it's pretty unlikely that you're going to fill the vacant 1GB of capacity anytime soon, and even if you did, the software will allow for a form of library handling. Let's be honest though, I can't see that happening!

Numa X Piano patches consist of four different samples, which can be layered or zoned across the keyboard. You could have a Yamaha Grand along with a Rhodes, across the entire keyboard, with a bass sound in the lower two octaves of the keyboard. It's flexible, but navigating the OS does take a little bit of time to get used to, although SL say that they are working on further shortcuts to make the process more streamlined.

The central OLED coloured display is the source of all operational information, presenting values succinctly, while reacting to any button press or pot-turn that you perform. To the left of the display, four infinite pots and accompanying buttons provide quick access to the editing process. At the upper-most editing level, these provide a route to select the part you want to edit, while pressing the associated pot down toggles the part on or off. This means that you can add or mute parts on the fly, which could be really handy for, say, adding strings mid-performance. The rotary element of the pots relates to part volume, so you have all four sample parts immediately open to switching and volume change, direct from the panel.

Pressing and holding the button that relates to a part immediately moves you into the next layer of editing, which provides the most useful and commonplace parameters. These change according to the current sample, so while a piano sample allows for alteration of tone and string resonance, a synth sound will offer filter cutoff and resonance. It's all very logically applied and very easy to access.

I tried my utmost to exhaust the polyphony on this instrument. The number of notes which are available at any one time is dependent on the number of samples in play, so if you're using a two-sample patch, polyphony halves. Having said that, »

Studiologic
Numa X Piano GT
£1799

PROS

- **The new Grand Touch keyboard action is sublime.**
- **Extensive selection of all the pianos and electrics you could need.**
- **Plenty of additional sample content.**
- **The audio I/O capability is exceptionally useful.**
- **As a playing experience, it's nothing short of a delight.**

CONS

- **Like all 88-note weighted keyboards, it's fairly heavy!**
- **Pitch and mod joysticks won't appeal to everyone.**
- **Menu navigation takes a little getting used to.**

SUMMARY

This is a beautiful keyboard which will appeal to any keyboardist or player with pianistic tendencies. The extensive included sample collection, with options for audio I/O, provides a comprehensive feature set within a tidy and uncluttered stage-piano format.

Little Brothers

While Studiologic were kind enough to send me their flagship Numa X Piano GT, which benefits from their top-flight weighted action, they are also producing the exact same piano in two lighter versions: the 88-note Numa X 88 (£1439) and the 73-note Numa X 73 (£1139). These lighter models employ Fatar's new TP-110 action, which is in turn an improvement on the older TP-100 action, found in many other high-end keyboards. Being lighter in weight, the action also benefits from a slightly speedier response. Other than the action, the functionality remains identical, with the only aesthetic difference being the absence of the wooden end-cheeks included on the GT model. All of the same benefits, but in an altogether lighter keyboard for easier transportation and gigging. And to further aid transportation, Studiologic make keyboard cases on wheels to accompany each model, available at an additional cost.

» Studiologic tell me that the total note count is well in excess of 200, so it's very unlikely that you'll be note-stealing any time soon.

Adding Externals

The team behind the Numa X Piano are highly competent musicians, who understand what it's like to play in live settings, and are clearly well versed with some of the issues that confront performing musicians on a day-to-day basis. Hence, the range is equipped with many useful features, such as a four-channel, on-board mixer, with the added ability to function as a USB MIDI and Audio interface. So if you want to drive, say, an outboard synth module without an additional mixer or interface, the Numa X Piano solves this problem immediately by allowing a module's audio output to be routed back in, blended at source, before heading back into the real world via the piano's main audio outputs. The capacity is also available for applying effects at this stage, and there's also a three-stage EQ, for basic tone-levelling adjustment.

The four audio physical inputs are accessed from the rear panel as unbalanced quarter-inch jacks, which may also be paired in stereo. This means that it's a total cinch to adjust while playing, or indeed mute up a channel on the fly. Another party trick allows the connection of a dynamic microphone, just like a mixer. As the audio inputs maintain access to the master effects section, you could feed a vocalist or even other instrumentalists through the piano, balancing levels, tone and effects on board. The signal sum then arrives at the stereo output from the keyboard. Genius!

If you also factor in the class-compliant USB audio element, this piano perfectly complements laptop use in a live setting too, with its own dedicated level control. Once you've finished your gig, take the Numa X Piano home, and you've got yourself a beautiful controller keyboard with audio interface capability, for a small home studio experience.

Gig Bound

During my brief time with the Numa X Piano, I had to conclude that this is a very desirable multifaceted powerhouse. Firstly, and quite importantly for any serious player, the keyboard is impeccable. It feels natural and, surprisingly for a fully weighted keyboard, rather effortless to play. The sample content is generous in the areas that you might expect, with beautiful acoustic and electric pianos at every turn, and a huge collection of other sounds which are very usable. It's got substantial capacity to effect samples on board, with all the associated effects one might associate with a Rhodes, Hammond or synth — and the flexible interfacing and patch assignment that allows any serious live player to set up patches to suit their way of working, adding signals externally, if so desired. The only element that didn't quite gel with me is the use of joysticks, for pitch, modulation or MIDI CC control. These take a little getting used to, and are a departure for players who may be used to pitch and mod wheels. Practice, as they say, makes perfect, and with more samples to follow very soon, the Numa X Piano GT has plenty of capacity for now, and the future.

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NEIL ROGERS

Kerwax Replica

Dual-channel Saturation Processor

Derived from the custom valve console in Brittany's famous Kerwax studio, the Replica harks back to a bygone age.

Kerwax is the name of a classy, all-analogue recording studio located in a former boarding school in Brittany, France. Owned by Christophe Chavanon, the studio pays homage to classic recording equipment of the 1940s to the 1970s, with the centrepiece now being a large, bespoke, wraparound valve mixing desk that was developed by the Kerwax team. (If you'd like to know more about Kerwax, check out our 2015 Studio File article: http://sosm.ag/studio-file-kerwax).

Now, they also make the Kerwax Replica that's reviewed here, and this device puts a pair of channels from their unique desk into an imposing 5U rackmount chassis. As well as offering two valve preamplifier channels, it features some intriguing creative tools for manipulating your audio signals.

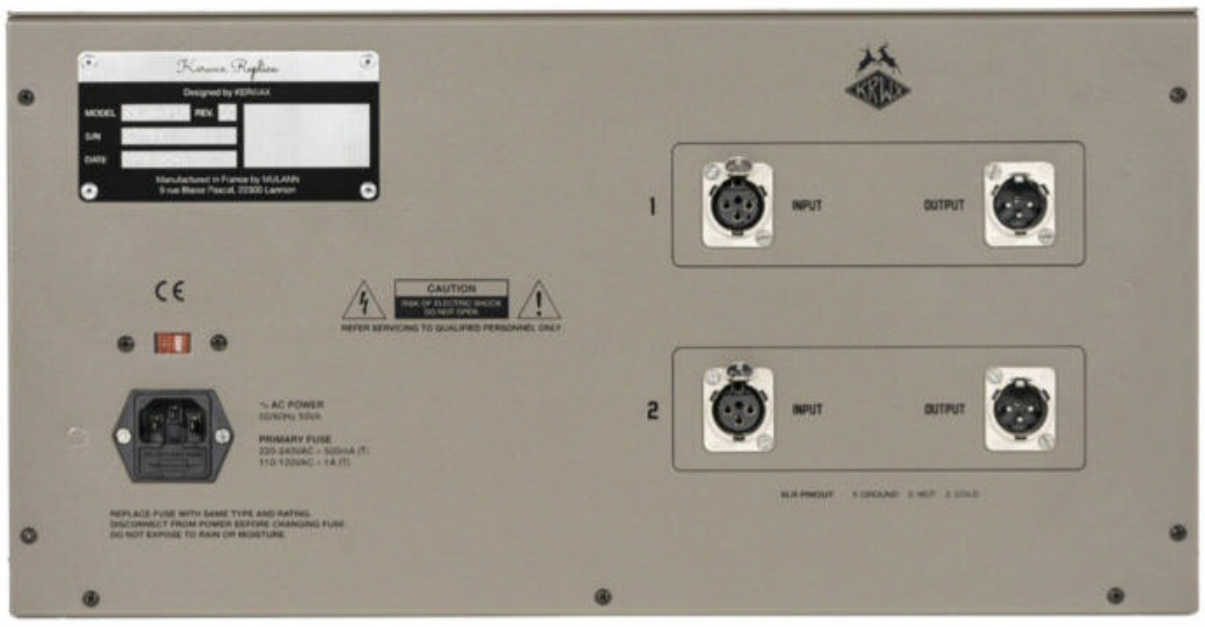
Features

The Kerwax Replica is a line-level processor, seemingly intended primarily as a device for running stereo mixes or submixes through, to introduce some of the characterful side-effects of vintage-style valve amplification. Each channel has completely independent controls, however, so you certainly aren't limited to stereo applications when recording and mixing. You have the option of running the Replica as a dual mono device or, intriguingly, in M-S mode, with the left channel becoming your Mid and the right controlling the Sides.

The Replica has 18dB of 'trim' available at the input stage, as well as a polarity reverse switch for each channel. The output level is displayed on large backlit VU meters, with the valves on display in a recessed 'shelf' between them. There's

a high-pass filter too, offering the options of cutting frequencies below either 80Hz or 100Hz, as well as a simple EQ section that allows for cutting or boosting high and low frequencies with Baxandall-style filters. Also available is a 'bias' control which allows for a little extra manipulation of the valves' behaviour. Each channel uses two 12AX7 valves with one used at the preamplifier and EQ stages and one for the 'drive' section — which can be switched in and out of the circuit. Last but not least, the unit features a substantial bypass/mute switch for each channel.

The Replica arrived at my studio in a large plywood box that set the tone for a product which screams 'attention to detail'. Occupying 5U of rack space, it is a formidable-looking, yet also incredibly stylish, piece of hardware. Some might suggest that the valves being visible on the front might look a little gimmicky, but



The obligatory rear-panel pic, with switchable IEC mains inlet and balanced XLR inputs and outputs.

there are practical reasons for having the valves outside of the unit, so why not? As well as ensuring heat from the valves can be easily dissipated, Kerwax encourage users to experiment with the subtly different character offered by different valves in the 12AX7 family.

In Use

The large, vintage-style knobs and all of the switches feel extremely satisfying in use and, despite the minor grumble that I found it hard to read the labelling on the EQ and bias sections in particular, I wasted very little time in getting the Replica patched into a hardware insert on my studio's patchbay.

I threw the Replica in at the deep end, patching it into my mix bus for the track I was working on at the time, which featured live drums, bass, guitars and piano. I began with the drive and EQ sections bypassed and, once I'd slightly backed off the level of the mix going into the Replica, I was impressed with how the track appeared just that tiny bit more exciting in the 4-6 kHz range. I could also clearly hear a pleasant 'tucking in' of the low-mids, and it sounded as if there was some inherent high-pass filtering being applied, with the very low frequencies on the bass guitar feeling a touch more restrained. I lost my nerve slightly when engaging the drive section, as the cymbals, in particular, started to sound a bit too distorted and I found the EQ section didn't offer anything obviously beneficial on this particular mix.

It was on a tracking session that I got to use the unit next, and this saw me working with a band over the course of several days. I was keen to have a consistent room sound for this project, to help knit together the diverse selection of instruments I would be recording. To this end, I had a spaced pair of omnidirectional mics positioned in the middle of the live room and I inserted the Kerwax across these mics for the duration of the session. It worked wonderfully, adding a subtle gloss to the room mics that got more characterful — almost compressed-sounding — on louder sources such as drums and bagpipes (yes, really!). I found myself experimenting with the EQ section a bit more here, as my confidence in the sound of the unit grew and I really appreciated that I was able to back off the top end slightly or add a bit more body via the broad shelving filters.

The Kerwax certainly has something to offer when recording live musicians »

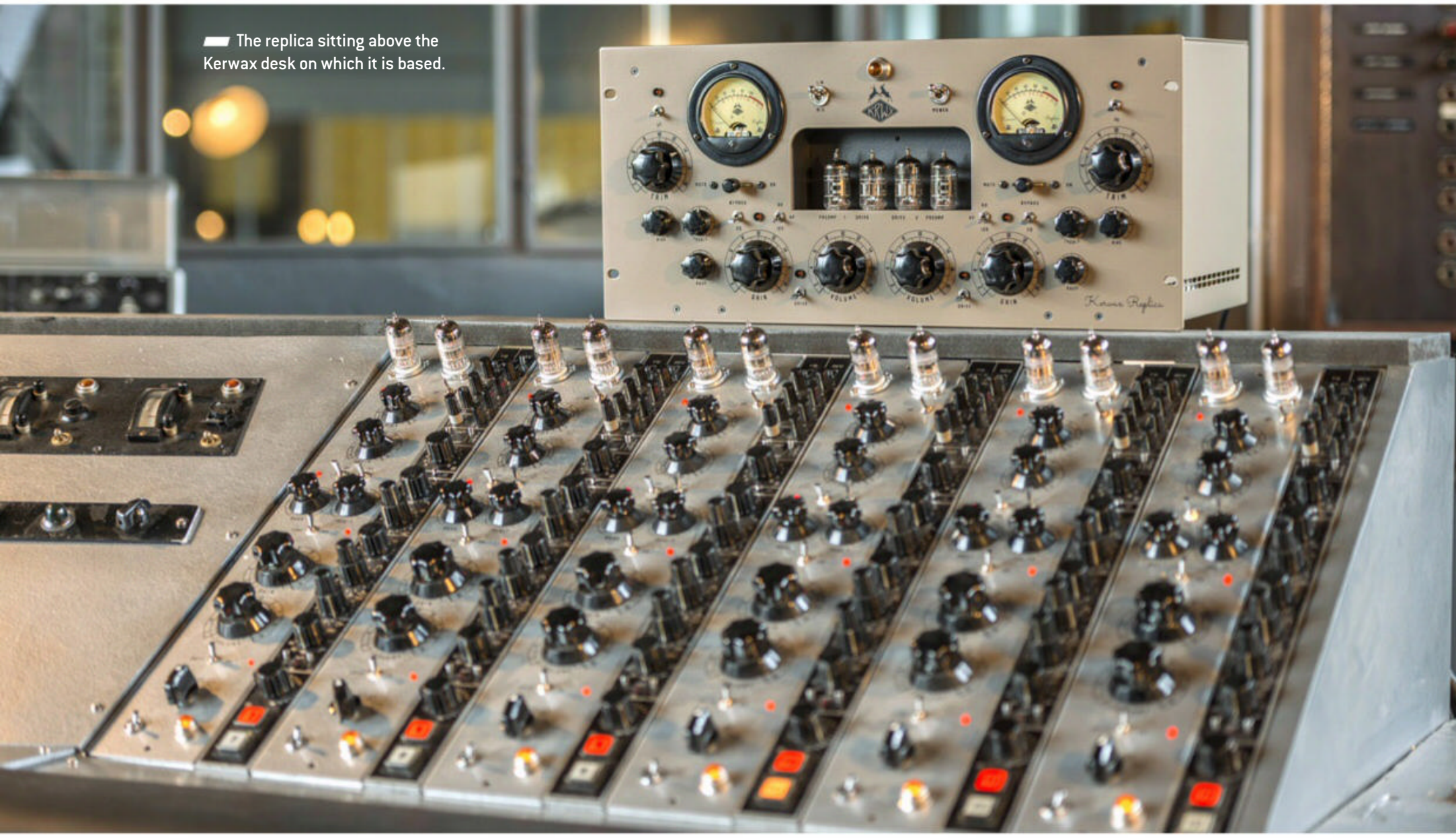
Kerwax Replica
£2699

PROS
- Impressive looks and styling.
- Excellent build quality.
- Great-sounding line-level valve amplification.
- Drive section adds real character.
- Useful filters, shelving EQ and M-S processing options.
- Relatively good value.

CONS
- Reading the labelling mid-session isn't easy.
- EQ perhaps didn't quite offer as much as I'd hoped when used on its own.

SUMMARY

The Kerwax Replica offers two channels of line-level amplification 'lifted' from the bespoke large-format mixing desk in the designers' own studio in Brittany, France. A substantial and impressively well-built device, the Replica offers a high-quality valve signal path, along with several useful additional features for imparting character to your tracks.



The replica sitting above the Kerwax desk on which it is based.

» but I actually found that it had a more substantial impact on proceedings when either mixing or creating programmed electronic or sample-based material. I loved what it did to a simple programmed drum pattern I was putting together for a track, and the drive section in particular imparted a lovely saturated effect that was well worth the extra effort of patching in a few cables and going outside the box. This became a common theme, in fact, and I found that any soft-synth part sounded more exciting when I ran it through the Kerwax with the drive section cranked up as far as I dared.

The EQ and bias controls are intended to allow the user more control with shaping the tone of sources as they react with the valve stages. I found the EQ section to be useful in this regard, especially when using the drive section in more of an extreme way. I didn't personally find them to add any magic when used as more of a traditional, conservative EQ on a mix bus, though, and found it was a similar story when playing with the bias control — it certainly offered some control but was often so subtle that I didn't really dwell on it.

I also spent quite a while auditioning the Replica in M-S mode. It's certainly a useful function, though I often found the results unnervingly dramatic if used on a whole mix. When used for shaping sounds, however, it's a very creative tool that can change the emphasis of synth parts or drum loops; it's a feature I could see myself using more of over time.

Final Thoughts

The Kerwax Replica makes a striking visual impression in a small studio; more than one client asked what 'that thing' did. But it's not merely a talking piece. The characterful drive section and creative tone-shaping combine to create a tool you could happily use at the writing, performance and production stages, before dialling things back and patching it into your master bus when mixing too, where it can help to knit things together.

The best of the plug-ins that model analogue saturation tools are seriously convincing these days, and while the differences are real, it's always a tough call to judge whether the often small gains that beautifully designed hardware such as this can impart are genuinely worth the extra investment. I won't tackle that question definitively here, but my impressions of working with this lovingly conceived and beautifully engineered unit are nonetheless worth stating. The differences were most obvious to me when I used it to process more electronic and sample-based music. Indeed, if you're an electronic producer looking for a flagship piece of hardware to add extra mojo to your productions, the Kerwax could well be precisely what you're looking for. Don't get me wrong, I also really liked the subtle valve sheen that it added to acoustic instruments and on a mix bus, but the differences in those scenarios seemed less stark to me, and I'd suggest that it falls more into the 'pleasant luxury' than the 'must have' camp. All in all, I couldn't help but be impressed with the Kerwax Replica and, considering its size and quality, the price isn't excessive for a professional tool, even if it's not exactly in home-studio territory.

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Mojave MA-37

Valve Capacitor Microphone

Mojave's new valve mic pays tribute to a neglected classic.

Mojave MA-37

£3499

PROS

- Wonderfully smooth sound.
- Attractive looks and high-class construction.
- Excels in roles where you wouldn't usually use a large-diaphragm valve mic.
- Lots of headroom for high-SPL sounds.
- Extended low-frequency response.

CONS

- Low sensitivity.
- Pattern setting is hard to confirm visually.

SUMMARY

David Royer's homage to the classic Sony C-37A shows exactly why that mic is revered.

SAM INGLIS

The classic valve mics we all know, love and wish we'd bought before prices went through the roof are mostly European. Mics like the Neumann U47 have become Platonic ideals, eternally out of reach and forcing us mortals to make do with shadowy imitations and inferior copies. Yet even before Neumann discontinued the U47, it was already facing serious competition from a surprising source.

Out Of The East

Like many national broadcasters, the Japanese NHK were heavily involved in developing audio technology after the Second World War. Ribbon mics from this period sometimes surface in the West, and are usually close copies of RCA models, but the first Japanese capacitor mic to enjoy major export sales was a more original design.

Created by NHK engineer Heitaro Nakajima along with Kanane Nakatsuru from Sony, the Sony C-37A resembled the U47 insofar as it was a large-diaphragm capacitor mic with a valve-based impedance converter. However, Sony took a very different approach from European manufacturers. The classic Neumann and AKG large-diaphragm capsules are all Braunmühl-Weber designs, in which two tensioned diaphragms and backplates are positioned back to back. These backplates are drilled with complex patterns of holes to create an acoustic labyrinth. Each side of the capsule can provide cardioid pickup, and when both diaphragms are energised, their respective contributions can be combined in the mic's electronics to synthesize other polar patterns such as omni or figure-8.

Sony's design, by contrast, used only a single diaphragm, with a much simpler backplate. Despite this, it still offered a choice of cardioid or omnidirectional polar patterns, thanks to a mechanical shutter which was operated by inserting a screwdriver into the back of the capsule. In omni mode, this blocked off a vent in the rear of the capsule and turned the C-37A into a pure pressure mic.

There have been small-diaphragm mics that used mechanical pattern switching, including several Schoeps models and the Shure KSM141. The same principle was also employed in the Philips EL6033/10, an obscure moving-coil dynamic mic, while Sony's use of it was possibly inspired by the RCA 77 ribbon mic. As far as I know, however, the C-37A and its derivatives are the only large-diaphragm capacitor mics that operate this way.

Sony's distinctive design approach also extended to the mic's impedance converter, a 6AU6 pentode valve wired as a triode in a cathode follower circuit. In this configuration, the valve can supply current gain but not voltage gain. This means the output level of a cathode-follower mic is typically very low, but on the plus side, it allows the output transformer and other electronics to be located at the other end of the mic cable, within the PSU. For this reason, European mic manufacturers tended to employ cathode-follower circuits mainly where there was a need for the mic to be as small as possible, such as in the AKG C60 and C12A.

This choice on Sony's part seems, by accident or design, to have helped the C-37A catch on in the US studio market. At its launch in the late '50s, most American studios were still using consoles and preamps created to work with passive ribbon mics. With close-miking coming into vogue, the C-37A's low sensitivity and high headroom made it easier to handle than the hotter European mics. Its relative simplicity also made it competitive on price.

Rediscovery

Though it never seems to have made inroads in Europe, the C-37A achieved widespread popularity in the USA, where it was used on countless classic recordings. Other Sony models also enjoyed some vogue through the '60s and '70s, but Neumann and AKG eventually reasserted their market dominance. Perhaps as a result, the C-37A remained something of a 'sleeper' while other valve mics such as the U47 and C12 were being rediscovered in the '90s. This particular sleeper, however, has long since been roused from slumber, and if you can find an original C-37A for sale, it will now set you back many thousands.

One of the reasons why C-37A prices are so high is that, until recently, no-one was making a modern recreation. The Chinese suppliers who churn out K47 and K67 copies have yet to produce a version of Sony's distinctive single-diaphragm design, and of the Western high-end specialists, only David Josephson seems to have tackled this particular capsule.

This situation may be about to change, however, thanks to another David. Most of us associate David Royer with ribbon microphones, thanks to the company that bears his name, but he has also designed many capacitor microphones, both valve and solid-state. Marketed under the brand Mojave Audio, these often draw inspiration from vintage models, but for the most part, they are not clones or remakes. The new MA-37 is, to some extent, an exception to this rule. David has admired the C-37A for many years, and in the MA-37, he's set out to produce a "modern take" on this unassuming classic.

Out Of The Box

The MA-37 ships in a large and robust Peli-style case. The mic itself is relatively compact, at about six inches long by two across, but the power supply is a hefty beast. A good-quality 5m cable connects the two, with the help of a short captive cable emerging from the mic. Like the Sony original, the MA-37 has an integrated yoke mount which attaches directly to a mic stand, in this case using a 3/8-inch thread. This doesn't provide much shockmounting, but generally makes positioning the mic quick and easy.

The US-made capsule at the heart of the MA-37 faithfully recreates the original, including the mechanical pattern switching. The mic's headbasket and body shape are also closely modelled on the C-37, but with the distinctive addition of small pegboard-style perforations all over the body. These provide some ventilation for the valve as well as making a distinctive visual statement. The overall build quality seems very good indeed, and the entire system oozes class in a nicely understated way.

The 6AU6 was never a particularly high-grade valve, and supplying low-noise examples today would be difficult, so the Mojave MA-37 uses an EF86 instead. The basic circuit topology is similar, however, with the Lundahl output transformer mounted in the PSU. Sony's own power supply went through a number of design iterations offering various different low- and high-cut options. For the MA-37, Mojave have settled on a three-way rotary switch

»

The large and weighty PSU incorporates both the output transformer and the high-pass filter.

» labelled M, V1 and V2. In the M (music) position the low end is allowed through unmolested, while the V (voice) positions introduce bass-cut filters with 6dB attenuation at 40 and 100 Hz respectively.

In the M setting, the MA-37's frequency response is specified as 30Hz to 18kHz within a ±3dB tolerance. It's not a mic you'd choose for recording bats, but those figures suggest a slightly greater high-frequency extension compared with the C-37A. Interestingly, David also says that the Sony capsule has a much greater bass extension than the classic Neumann or AKG large-diaphragm designs. The frequency response chart bears this out, with the mic in cardioid mode being only 5dB or so down at 20Hz. In omni mode, the plot shows a low shelving boost that begins around 700Hz and is still 2dB up even at 20Hz. Elsewhere, the published response in both modes is very similar, being flat through the mid-range and rising to a gentle, broad peak around 12kHz.

The MA-37 can accommodate SPLs of up to 135dB before 0.5 percent distortion is reached, and self-noise is specified at 18dB or better. At 5mV/Pa, its sensitivity is more akin to moving-coil mics than capacitor models; by way of comparison, the current Neumann U87Ai offers 28mV/Pa in cardioid mode, and the Shure SM57 clocks in at 1.7mV/Pa.

Other Options

If, like me, you've fallen in love with the C-37A sound, at least as represented by the MA-37, there is one obvious alternative. The Tonelux JC-37 was developed in conjunction with renowned producer and C-37A fan Joe Chiccarelli, and recreates both the appearance and the design of the Sony original. Unlike Mojave, however, Tonelux have chosen to simplify the mic and PSU, so the JC-37 has a fixed cardioid pattern and no bass cut filters. Tonelux say this gives the mic "a cleaner signal path", and if you fancy stereo C-37A goodness, a single Tonelux PSU can power two JC-37 mics.

David Josephson has also long been a fan of the C-37A, and used it as the inspiration for the original Groove Tubes mics back in the '90s. These are occasionally available second-hand, but be careful what you're buying, as later models used a different capsule. Josephson's Sony-style capsule is used in the current C705 and C715 mics, though these are far from being straightforward C-37A clones. Single-sided large-diaphragm capsules are also employed by Samar Audio in the TF10 and JZ in the BB29. Both are fine mics, but neither sounds remotely like a C-37A.

Finally, Sony themselves redeployed the same capsule in other mics after the C-37A was discontinued. These include the solid-state C-37P, but also valve models such as the C-47 and the original C-800, not to be confused with the famous C-800G. Most of these valve mics are, if anything, even rarer outside Japan than the C-37A.

The Sony Sound

The Sony C-37A is a rare mic in the United Kingdom. I've never seen one in any British studio, even Abbey Road, and wasn't able to get hold of one for comparison. So I can't tell you whether the MA-37 sounds exactly like the C-37A, but that is perhaps beside the point. It's not intended as an obsessive copy of that mic, more as an attempt to create a new mic that shares its good qualities. So it's encouraging that the clichés that sprang to mind when I tried it exactly matched the clichés that are widely shared regarding the C-37A.

If I had to pick one adjective above all to describe the sound of this microphone, it would be 'smooth', and that's much more than just a function of its relatively neutral frequency response. There are plenty of flat mics out there, but the MA-37 isn't merely flat. On the one hand, it captures a nicely balanced sound, and at sensible working distances, doesn't come across as muffled or veiled; yet, on the other, it's mysteriously deaf to unwanted resonances and harshness. It's almost as though it incorporated some sort of sophisticated dynamic equaliser that can seek out and minimise edginess »

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Switching patterns on the MA37 requires a screwdriver and a deep breath...

» in the source. (Given its extended bass response, it's also surprisingly resistant to plosive pops.)

It's not a universally great vocal mic — the sort of voice that is flattered by a U87 or similar mic can sound a little pillowy with the MA-37 — but it is a really useful option for those 'difficult' voices that don't work well on other capacitor mics, especially female vocals with a nasal quality or a tendency towards screechiness. In cardioid mode, it also has quite a bit of proximity effect, which you can use to your advantage on thin-sounding vocalists. I didn't get the chance to try it on brass instruments, but I imagine it would make a fantastic close mic for trumpets and their ilk.

Where I really loved the MA-37, however, was on sources where you perhaps wouldn't usually consider using a large-diaphragm valve capacitor mic at all. It feels all wrong to position a mic like this right up against a snare drum, for example — but overcome your squeamishness, and the results can be spectacular. There's none of the usual emphasis on ringing and grittiness; instead, you get a soft, inviting bloom in the low mids, with a gorgeous 'puffy' quality at the top end. And my favourite application of all was on guitar amps. I'm not one of those people who's obsessive about mic choices for electric guitar, but I was really surprised by how different the MA-37 sounds from other mics I'm familiar with. It de-emphasises the scratchy stuff in the upper midrange, delivering a richness and polished quality to the sound without making it boomy or congested. The only way I can fully describe the effect is to say that it makes the amp sound a lot more expensive than it really is!

Open The Shutters

Talking of squeamishness, I imagine most of us will take a deep breath before plunging a screwdriver into the back of a capsule to change the polar pattern. Assuming you have a suitable implement lying around, though, it's actually no harder than flicking a switch, and the only real downside is that there's no visual indication to tell you whether the mic is in cardioid or omni mode. As you'd expect, the omni pattern is pretty approximate; there's more acoustic shadowing going on than in a typical pencil mic or a conventional multipattern capacitor model, and sound arriving from the rear loses a fair bit of top end. There are useful applications for omni mics that narrow to a more focused pattern at high frequencies — that's what is usually prescribed for a Decca Tree, for example — but in this case I suspect the omni setting might be more valuable for the on-axis tonal changes it brings, especially at distances where the proximity effect would otherwise be apparent. On guitar amps, for example, the omni pattern sometimes seemed to deliver a thicker low midrange. I can also imagine using it off-axis in omni mode to tame a particularly bright or harsh source.

Inasmuch as the MA-37 has any negatives, they're ones it shares with the Sony original. The biggest is probably the mic's low sensitivity. This can be helpful in high-SPL situations, but it can raise issues too. Naturally, the mic uses a good-quality screened cable, but even so, a small amount of hum pickup was audible in my studio space when I wanted to use it on quiet sources. This isn't a criticism of the mic, just a caution that running a very small signal voltage through a relatively long cable can make you newly aware of a less-than-perfect electromagnetic environment! Some physical hum was also audible from the power transformer, though it wouldn't be enough to bother me in practice. In terms of self-noise, the review mic was more than quiet enough for general studio use, and I suspect quieter than most original C-37As.

All things considered, then, this is perhaps not the ideal first mic, or only mic. But, again, that's not intended as a criticism. What makes it great is that however many mics you already have, the MA-37 will bring something genuinely different to the party (unless you're lucky enough to own a C-37A, of course). And unlike many 'interesting' or 'different' mics, it's truly useful on a wide variety of sources. I'd even go so far as to say that on some of them, it does things you won't get from any other type of mic.

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Apple MacBook Pro

Laptop Computer

Apple's most powerful laptop computer ever has been explicitly designed with music in mind.

MARK WHERRY

October 2021 marks the 30th anniversary of Apple's first notebook computers. The PowerBook 100, 140 and 170 helped set the template for the modern laptop, incorporating now-cornerstone design features like the palm rest below the keyboard. IBM would refine this approach with the help of German-born designer Richard Sapper, launching the ThinkPad a year later, and the rest is history.

The latest generation of MacBook Pros are perhaps equally revolutionary, being the first notebooks Apple have designed from the ground up to fully embrace their latest custom silicon. What's more, nearly every Apple executive featured in the launch presentation emphasised the important role music plays at the company. Apple's CEO, Tim Cook, even began the keynote by saying the event was about two things: "Macs and music".

Apple MacBook Pro
From £1899

PROS

- A serious piece of kit, engineered to perfection with a display to die for.
- The M1 Pro and Max chips are designed for peak notebook performance.
- Physical function keys are back!
- Some will welcome the return of the MagSafe connector, as well as the HDMI output and SDXC card reader.

CONS

- The System-in-Package nature of the M1 family's design means memory and processing capabilities can't be configured separately, which can have an impact on the cost.
- The notch isn't exactly a welcome inclusion in a Mac.
- Wired Ethernet in the PSU would have been nice.

SUMMARY

The highly anticipated, next-generation MacBook Pros, featuring the new Pro and Max iterations of Apple's custom M1 chip, are pretty much everything you could wish for in a notebook. Available with the same specifications in either a 14- or 16-inch design, the new MacBook Pros offers a gorgeous screen, fast storage, improved battery life, and re-implementation of certain older keyboard facilities and connections that had previously been deprecated.

Good Looks

The sleek design of the new MacBook Pro is, as many have remarked, reminiscent of the first 'Titanium' PowerBook G4 that debuted in 2001. And as always, the MacBook Pro's precisely machined bodywork is part of an "advanced thermal system" that makes it possible to achieve sustained performance without fan noise. The 16-inch model is almost but not quite the same size as the previous Intel MacBook, but at 4.8lb (2.2kg) to the older machine's 4.3lb (2kg), it's a little heavier. However, if you want the same features in a slightly lighter and smaller form factor, Apple also offer a new 14-inch MacBook Pro. This weighs in at 3.5lbs (1.6kg) with a height of 1.55cm and a size of 31.26 x 22.12 cm. And, for the first time, the 14-inch model can be configured with the same specifications as the larger model.

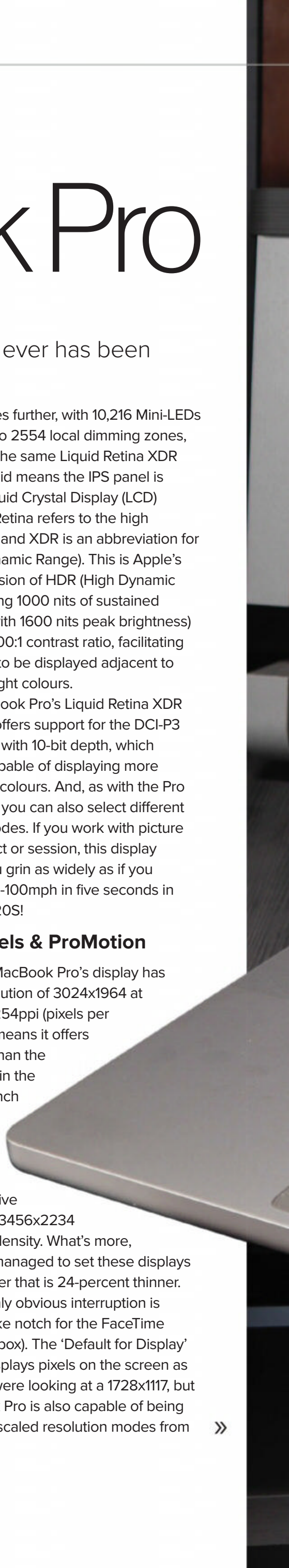
Apple have a justifiably high reputation when it comes to displays, and there probably aren't enough superlatives in the English language to describe this one. Built upon and leapfrogging the technology used in both the high-end Pro Display XDR, and the Liquid Retina XDR display used in the latest 12.9-inch iPad Pro, it really is stunningly good.

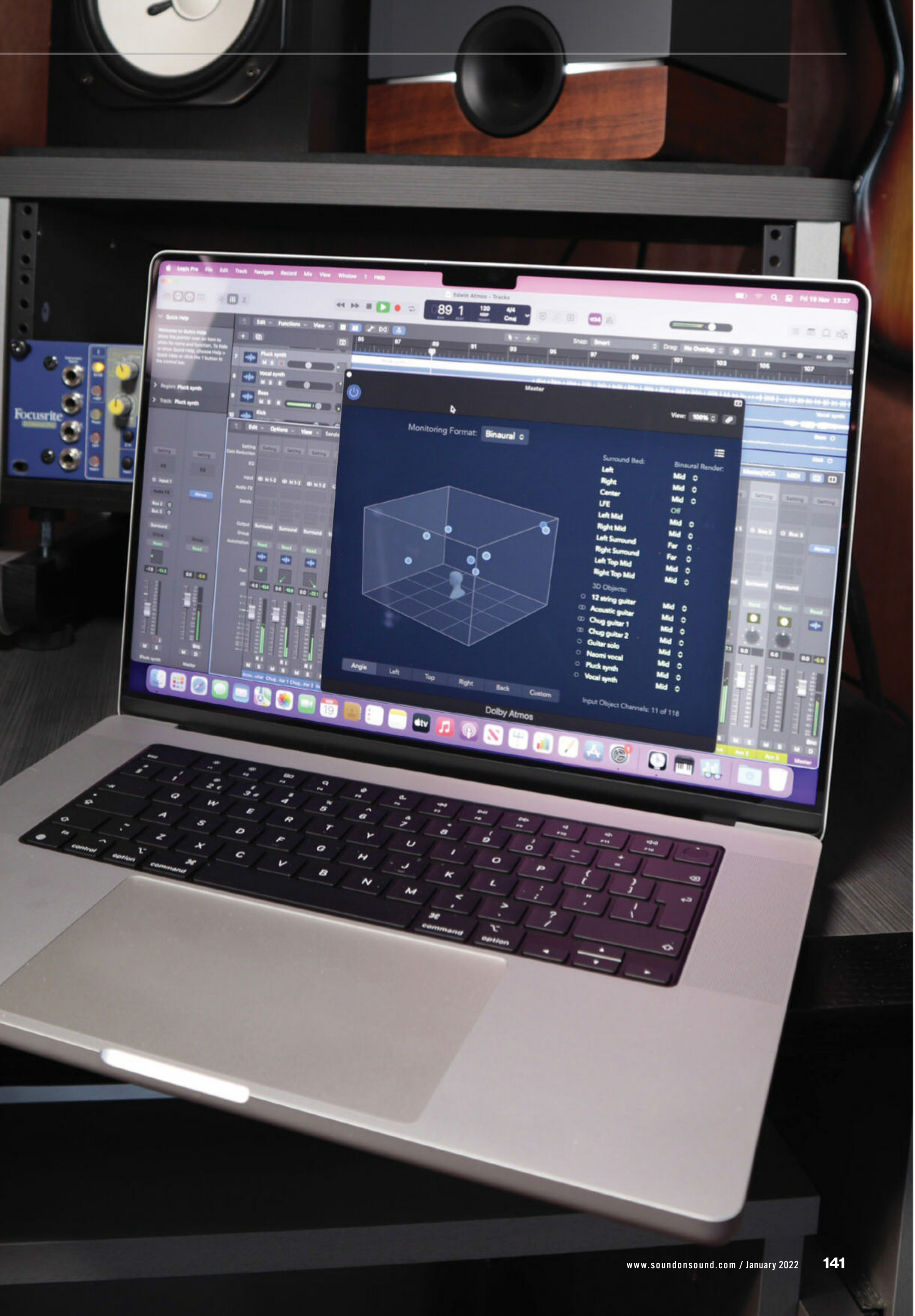
The Pro Display XDR was released in 2019, and uses a 2D backlight system comprising an impressive 576 LEDs with 576 corresponding full-array local dimming zones, managed by a custom, Apple-designed timing controller. Apple adapted the technology for the latest 12.9-inch iPad Pro, resulting in the spectacular Liquid Retina XDR display. However, because the iPad Pro is considerably thinner than the Pro Display XDR, it was necessary to use Mini-LEDs for the backlight, making it possible to pack over 10,000 of these smaller light sources organised into more than 2500 dimming zones. The new MacBook Pro's 2D backlight goes further, with 10,216 Mini-LEDs organised into 2554 local dimming zones, but also has the same Liquid Retina XDR moniker. Liquid means the IPS panel is based on Liquid Crystal Display (LCD) technology, Retina refers to the high pixel density, and XDR is an abbreviation for (Extreme Dynamic Range). This is Apple's 'extreme' version of HDR (High Dynamic Range) offering 1000 nits of sustained brightness (with 1600 nits peak brightness) and a 1,000,100:1 contrast ratio, facilitating deep blacks to be displayed adjacent to gloriously bright colours.

The MacBook Pro's Liquid Retina XDR display also offers support for the DCI-P3 colour space with 10-bit depth, which means it's capable of displaying more than a billion colours. And, as with the Pro Display XDR, you can also select different reference modes. If you work with picture in your project or session, this display will make you grin as widely as if you were going 0-100mph in five seconds in a McLaren 720S!

Pixels & ProMotion

The 14-inch MacBook Pro's display has a native resolution of 3024x1964 at a density of 254ppi (pixels per inch), which means it offers more pixels than the display used in the previous 16-inch MacBook Pro; the new 16-inch MacBook Pro has a native resolution of 3456x2234 at the same density. What's more, Apple have managed to set these displays within a border that is 24-percent thinner. In fact, the only obvious interruption is the iPhone-like notch for the FaceTime camera (see box). The 'Default for Display' resolution displays pixels on the screen as though you were looking at a 1728x1117, but the MacBook Pro is also capable of being used in four, scaled resolution modes from »

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Surround Bed:
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Right Mid
Left Surround
Right Surround
Left Top Mid
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3D Objects:
12 string guitar
Acoustic guitar
Chug guitar 1
Chug guitar 2
Guitar solo
Naomi vocal
Pluck synth
Vocal synth
Angle
Left
Top
Right
Back
Custom
Input Object Channels: 11 of 118
Dolby Atmos
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Not The Notch?

If you're an iPhone user, you'll already be familiar with the notch: an intrusive, small black rectangle that makes an unwelcome appearance in the top of the display. This is how Apple have incorporated the new 1080p FaceTime HD camera in the MacBooks. It's a big improvement over the previous 720p equivalents from the Mesozoic era, although still not as stunning as, say, the iPhone Pro's front-facing TrueDepth camera. As has become the norm, the video from the built-in camera is enhanced via Computational Video courtesy of the M1 Pro or Max's Image Signal Processor. But why Apple don't just bring the 'Pro' camera from the iPhone Pro to the MacBook Pro remains something of a mystery.

Much like iOS on the iPhone, Mac OS Monterey adapts its user interface to accommodate the notch within the menu bar to give the illusion of seamlessness. This means that the size of the menu bar remains consistent, no matter which display scaling mode might be selected, but it also means that if an application's menu bar titles can't be accommodated to the left of the notch, Mac OS interrupts the menu bar before continuing to display the remaining titles to the right of this area.

The MacBook Pro sees the welcome return of connectors such as HDMI, and an SD card slot.

» 1168x755, 1312x848 and 1496x967 all the way up to 2056x1329, which is especially good for eagles.

To further enhance the new display, Apple have also introduced ProMotion from the iPad Pro and the most recent iPhone 13 Pro. To understand why ProMotion is beneficial, consider that most computer displays have a fixed refresh rate, which is typically 60Hz (basically, 60 frames per second). The new MacBook Pro display has a refresh rate of 120Hz (which, incidentally, is why it can support Dolby Vision), allowing for a user interface that feels extremely fluid. What ProMotion makes possible is the ability to set the display's frame rate adaptively, according to the amount of movement being rendered.

Relatively static content such as word processing doesn't require a rapid refresh rate, and reducing it can lower the energy requirements. By contrast, when you zoom in and out of waveform displays, the MacBook will increase that rate such that the on-screen motion appears smoother. And the quality of video playback goes without saying. Where ProMotion really shines in music and audio software is when you enable Logic's Scroll In Play feature, Cubase's Stationary Cursor Auto-Scroll mode, or Pro Tools' Continuous scrolling setting, allowing the edit windows to slickly glide along without the usual unevenness.

If you're working with video and need to lock the display to a specific frame rate instead, you can do so in the System Preferences' Display panel. The MacBook Pro's Liquid Retina XDR display offers five fixed refresh rates: 60Hz, 59.94Hz, 50Hz, 48Hz, 47.95Hz. As you may notice, these frequencies haven't been chosen by random: each setting is either exactly a 99.9-percent version (a hangover from NTSC colour) or double a commonly used frame rate, such as 60fps, 50fps, 48fps, 30fps, 29.97fps, 25fps, 24fps and, of course, 23.975fps.

Get Yourself Connected

This is usually the point in a Mac laptop review where I'd criticise the lack of expansion ports. However, this time, Apple have uncharacteristically resurrected some of them, including our old friend the MagSafe connector, now in its third, even smaller incarnation (see box). The HDMI port will be helpful for those working with video, while photography and video users will welcome the return of an SDXC slot; there are also many audio field recorders that make use of such memory cards. The slot itself supports the SD 4.0 standard and is compatible with UHS-I and UHS-II SDXC cards, but not UHS-III cards supporting SD 6.0 (which was released back in 2017) or the newer SDUC format.

The main interface for external expansion remains USB-C, with three connectors — two on the left side and one on the right — each supporting charging, DisplayPort, Thunderbolt 4 and USB4 (both at up to 40Gb/s). This is, of course, vital for connecting MIDI and audio interfaces, any video or additional screens, copy-protection devices, and external PCIe cards via a compatible chassis (see box). Using the HDMI and USB-C connectors, an M1 Pro MacBook Pro supports up to two Apple Pro Displays XDRs (or displays with up to a 6K resolution at 60Hz), whereas a MacBook Pro with an M1 Max can handle up to three Pro Display XDRs (or 6K displays at 60Hz) plus one 4K television or equivalent display.

The MacBook Pros have the usual 3.5mm headphone output you'd expect to see on a Mac, but this connector has also received some uncharacteristic professional love. The digital audio output facility that used to be part of the headphone jack (using a suitable Toslink cable) was dropped from the MacBook Pro in 2016. It's not back, and with support for sample rates up to 96kHz.

Even more useful for musicians and audio engineers, however, is that the headphone jack features DC load detection and adaptive voltage output, enabling the output to support line-level output as well as low- and high-impedance headphones. According to an Apple support document: "When you connect headphones with an impedance of less than 150Ω, the headphone jack provides up to 1.25 volts RMS. For headphones with an impedance of 150Ω to 1kΩ, the headphone jack delivers 3 volts RMS." With my trusty Sennheiser HD650s, the overall sound was noticeably cleaner than on previous Mac laptops, with an appropriate amount of amplitude for my sensitive ears. »

RIP Touch Bar, 2016-21

Apple have had a bit of a hit-and-miss relationship with notebook keyboards over the last few years, persisting with the loathsome Butterfly mechanism for much longer than most would have wished. With the new MacBook Pro, they have finally put the Touch Bar out of our misery and returned to physical function keys with a wider Escape key. While the Touch Bar looked neat, it seemed like a solution in search of a problem, and although there were a few clever uses for it, such as QuickTime presenting transport controls or GarageBand and Logic Pro offering a virtual piano keyboard, ultimately there weren't enough of these to make you overlook the omission of physical keys.

The Touch Bar is gone, replaced by physical function keys.

SOUND ON SOUND

VIDEO

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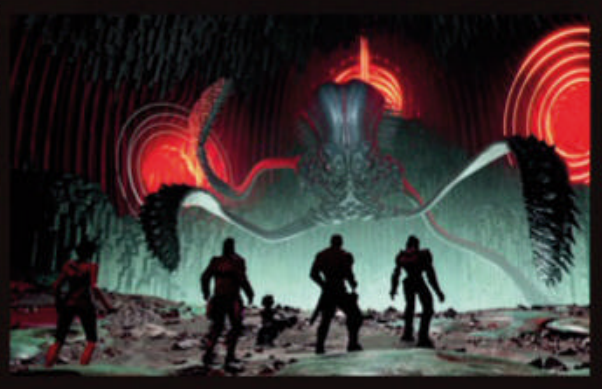
Richard Jacques:
Scoring The Video Game

The epic soundtrack to the *Guardians Of The Galaxy* video game is one of the biggest projects ever undertaken by a single composer. In our exclusive feature, Richard Jacques describes how he created nearly six hours of high-impact orchestral music — and shows us how the main theme developed from an initial MIDI demo to the final version recorded at Abbey Road.



www.youtube.com/soundonsoundvideo

» Audio professionals are unlikely to use built-in loudspeakers or mics, but it's worth noting that Apple have put a lot of work into developing them. An Apple-designed 'virtualiser' brings Spatial Audio support to a Mac notebook's speaker system for the first time, and the MacBook Pro's three-microphone array is perfect for everyday tasks like recording voice memos or participating in FaceTime calls. As luck would have it, whilst I was working on this part of the review, my housemate was playing an acoustic guitar on the sofa in the living room. Soon the new 16-inch MacBook Pro was perched next to him — and the results were actually pretty good, with a surprisingly low noise floor.

Finally, there's support for both 802.11ax Wi-Fi 6 wireless networking, which is IEEE 802.11a/b/g/n/ac compatible, and Bluetooth 5.0 wireless technology — although I wonder when Apple will start supporting newer 5.x specifications.

Core Strength

Before announcing the new MacBook Pro itself, Apple first introduced the next generation of Apple Silicon, an evolution from the original M1 chip released around a year prior. Where the original M1 featured four performance and four energy cores yielding a CPU with a total of eight cores, the new M1 Pro and Max chips each have two energy cores and eight performance cores. The only exception is the M1 Pro used in the base 14-inch model, which has six performance cores. Additional M1 Pro and Max chip variations are available featuring different numbers of GPU cores (see box).

Apple don't discuss clock speeds for these cores, and frankly there's no reason they should, as they don't really convey anything meaningful. What they do say is that the overall performance of the 10-core M1 Pro and Max chips is around 1.7 times that of the M1. In any case, performance isn't exactly an issue with the original M1 — quite the opposite, in fact! The biggest limitation is probably that chip's 16GB memory ceiling. While M1-based systems with 8 or 16 GB memory outperformed the assumptions one would usually have of computers with these specs, such amounts of physical memory clearly weren't enough for substantial projects, or projects heavily dependent on sample libraries.

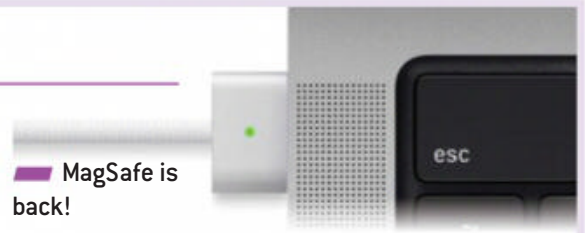
One of the reasons for this memory ceiling limit was because the memory chips were included in the M1 package itself, making it possible to reduce the latency associated with accessing memory. If more memory had been included, the cost would have been greater, and a larger, more power-hungry package would have been needed, adding to the challenge of repurposing the existing MacBook Air, 13-inch, and Mac mini enclosures used in the first Apple Silicon-based Mac products. However, the new MacBook Pros have been designed from the ground up for Apple's new latest silicon, and can accommodate both a larger System-on-Chip (SoC) and additional memory. The M1 Pro supports either 16 or 32 GB of memory, while the M1 Max goes further, offering 32 or 64 GB.

The way in which M-series silicon handles memory is vital to the overall performance of the system, with Apple adopting an architecture referred to by the company as unified memory. Simply put, unified memory is accessible by all the other components that comprise the System-on-a-Chip: from the CPU and

Power & Charging

The 14-inch MacBook Pro is powered by a 67W or 96W power adaptor depending on the configuration, whereas the 16-inch comes with a larger 140W adaptor. The adaptor itself has a USB-C connection and attaches to the MacBook Pro's dedicated MagSafe 3 power connector via a supplied 2m cable. The key feature of MagSafe is magnetic quick release, meaning that your laptop doesn't fly off the table if someone trips over the cable. USB-C connectors can also power and charge the MacBook Pro.

The 14-inch MacBook Pro has a 74W/h (Watt-hour) lithium-polymer battery rated at 69.6W/h, while the 16-inch model has a 100W/h lithium-polymer battery rated at 99.6W/h. This is especially relevant if you intend to take your MacBook Pro on a plane; both the TSA (Transport Security Administration) in the US, and the CAA (Civil Aviation Authority) in Britain specify 100W/h as the maximum battery capacity for portable electronic devices that can be taken on board as carry-on luggage.



MagSafe is back!

A new feature is Fast Charging, which makes it possible to go from zero to 50 percent in just 30 minutes. However, this only works if you have the right combination of power sources and cables. For those with a 16-inch MacBook Pro, you'll need to be powered via MagSafe 3 and the included 140W power adaptor, while Fast Charging is possible on the 14-inch with a 140W or 96W power adaptor via MagSafe 3, the 96W adaptor with a USB-C charge cable, or, alternatively, with a Pro Display XDR with a Thunderbolt 3 cable, or a third-party external display capable of delivering 94W with either a Thunderbolt 3 or USB-C cable. The new 24-inch, M1-based iMac offers support for wired Ethernet in the power supply, but sadly that's not an option here.

Storage Scores

The 16-inch MacBook Pro offers the fastest storage system yet seen on any Mac. In their tests, using the Flexible I/O tester (FIO 3.27), which is a comprehensive tool for performing simulated I/O workloads, Apple report a result of 7.4GB/s for pre-production 14- and 16-inch MacBook Pros powered by M1 Max chips with 64GB memory and an 8TB SSD. The small print specifies the test used a 150GB file, accessed via eight active (in-flight), 1024kB blocks. This is obviously rather impressive performance. And even running a slightly more mundane test — the ever-trusty AJA System Lite for consistency with tests from other reviews — the new 16-inch MacBook Pro yields read and write scores of 4198MB/s and 3675MB/s respectively. Both results are around 800-1000MB/s faster than the previous leading Macs, the Mac Pro and the last Intel-based MacBook Pro. AJA System Lite is still Intel-native, but given the nature of these tests, I don't think Rosetta 2 impeded the results in any significant way.

In order to achieve these results, the M1 Pro and Max feature a new, "advanced" storage controller that accesses fast NAND for the SSD, and you'll notice this gives notably better write speeds than the original M1-based Macs. Unified Memory is greatly helped by the speed of the internal storage system when paging blocks of memory in and out of the available RAM.

For musicians and audio engineers, it's not even worth considering the number of audio files or streaming sample voices that can be played. Trust me, whatever real-world number you're aiming for, the new MacBook Pros will be able to deliver the required I/O. The bottleneck now, as it's probably been for some time, is the software hosts and plug-ins themselves. Apple showed me a Logic Project as a demonstration which was loaded with many, many instances of Spitfire Audio's BBC Symphony Orchestra library playing back, and I don't think the MacBook Pro even started up the fans in the cooling system!

The new MacBook Pro starts with a 512GB SSD, which is rapidly becoming the new, comfortable minimum amount of storage you'd want in such a system. Some of the more expensive standard configurations are equipped with 1TB of storage, and any of the new MacBook Pros can be ordered with 1TB, 2TB, 4TB and 8TB SSD options, though the last of these will set you back another £2000, approximately.

GPU cores to the custom, domain-specific accelerators like the Neural Engine. The M1 Pro and Max support LPDDR5 SDRAM, whereas the M1 used LPDDR4X SDRAM. Consequently, the M1 Pro's memory bandwidth is nearly three times greater than that of the M1 at 200GB/s, with the M1 Max doubling this to 400GB/s.

Memory bandwidth is incredibly important to applications and plug-ins that work with audio streams. Because these kinds of data steams are generally contiguous, caches aren't particularly effective, meaning that memory is constantly being accessed directly. Bandwidth aside, the ability to once again have up to 64GB memory in a MacBook Pro is most definitely welcome; 128GB would have been even more welcome (higher-end Windows-based notebooks, such as certain workstation-oriented ThinkPads or ZBooks from the likes of Lenovo or HP, have offered this ability for a while), although isn't this always the case when it comes to memory?

However, configuring a MacBook Pro with lots of memory raises one particular problem with Apple's use of a System-in-Package (SiP) design. The number of CPU and GPU cores is inextricably linked with the amount of unified memory, so buying more memory means paying for graphics power you might not necessarily need.

The Engine Room

The M1 already had support for video encoding and decoding, providing dedicated hardware acceleration for H.264 and HEVC (H.265) codecs. In the M1 Pro and M1 Max chips, these features are part of a newly dubbed Media Engine, which also adds hardware acceleration for Apple's own ProRes and ProRes RAW codecs, along with an additional ProRes encode/decode engine. This is reminiscent of Apple's add-on Afterburner card for the Mac Pro, which utilised an FPGA to accelerate workflows involving Apple's ProRes and ProRes RAW codecs. Since FPGAs are programmed with logic defined by a hardware description language, it seemed possible this work could be reborn into fixed hardware within an M-series SoC. Apple wouldn't confirm or deny this speculation, although I got the distinct impression I saw a brief, dry grin from someone on the engineering side.

How, then, does the new MacBook perform? I think it's fair to say that modern high-end computers of all types are now so powerful that it's no longer useful to count the number of plug-in instances that can run simultaneously. If anyone is making real-world projects that require 1500 instances of Amp Designer then I probably don't want to hear them! I hope to be able to update our Mac content in future with a new and more relevant music-based benchmarking system, but in the meantime, Primate Labs' Geekbench remains the most universal way of comparing the overall performance of different machines. On the 16-inch MacBook Pro, its CPU tests reported a single-core score of 1778 and a multicore score of 12,738.

The first of these represents only a small improvement over existing M1 models, which typically score just over 1700 in the single-core tests — but that nevertheless makes it the highest single-core score recorded for any Mac, including the Mac Pro. And the multicore score leaves the other M1 models far behind. The M1 iMac and Mac mini turn in numbers around 7400, suggesting that Apple's claims of a 1.7x performance gain are accurate. To put that into context, the 16-inch Macbook Pro outperforms most configurations of the Intel iMac Pro, as well as base Mac Pro configurations. To gain substantially better performance, you'd need at least 16 cores in a Mac Pro, and that is not cheap!

Finally, one of the new MacBook Pro's greatest attributes for musicians and audio engineers is that, while the cooling system has a fan, you rarely hear any fan noise. During most recording tasks, the notebook will remain completely cool and therefore silent. This means it's perfectly possible to use the MacBook Pro in the same live room as the musician or musicians you're recording.

Closing The Books

The new MacBook Pro is, in many ways, everything you want it to be, although it's also fair to say this might come at a price you'd rather it not be. As ever, the starting prices for the base models are not unreasonable, but there's a fairly steep cost gradient as you add more memory and storage, and the fully loaded 16-inch 2021 model with an M1 Pro, 64MB memory and 8TB of storage tips the scales at £5899. However, as you might have been able to infer from the general tenor of this review, I can confidently say the new MacBook Pro is quite simply the best laptop I have ever used. In everything from its raffiné-yet-utilitarian design to the stunning display, and the pure potential of mobile performance, this thing is a beast!

£ From £1899 (14-inch) or £2399 (16-inch) including VAT.
W www.apple.com/uk/

Test Spec

- 16-inch MacBook Pro with an M1 Max, 64GB memory and 8TB storage.
- 14-inch MacBook Pro (base model) with an M1 Pro, 16GB memory and 512GB storage.
- Mac OS 12 Monterey, as installed by Apple.
- The 16-inch model was preloaded with Logic Pro 10.7 as well as many tools for video and 3D rendering, such as Final Cut Pro, Affinity and Cinema 4D.

It Had To Be GPU

Compared to applications such as AutoCAD, Cinema 4D, Da Vinci Resolve or Apple's own Final Cut Pro, the daily needs of music and audio software regarding graphics processing are relatively modest. Therefore, any of Apple's GPU configurations used in the M1 Pro and Max chips are likely to be more than enough for your needs, especially since video playback is handled by the Media Engine rather than burdening the GPU (or CPU).

Unlike the ARM architecture, which is licensed to implement Apple Silicon's CPU cores, the GPU architecture is of Apple's own design and thus relatively little is known officially about its specifications beyond a handful of performance claims made by Apple. However, what is clear, is that in order to take advantage of the graphics processing available to the M1 family, an application must embrace Apple's Metal. This essentially combines the functionality for OpenGL (for graphics) and OpenCL (for running domain-specific code on heterogenous systems, where more than one type of processor is used).

By utilising the Metal graphics API (application programming interface), users can enjoy faster rendering of user interface and data, and a good example of this was seen when Steinberg adopted the use of Metal in Cubase 11. The interface became significantly faster and smoother; prior to this, vertically scrolling the Project window with a large number of tracks to find just one was much like trying to find the dry bit in a bathroom's rotating towel dispenser.

To balance the differing visual needs of users, Apple differentiate M1 Pro and Max-based systems by including a different number of GPU cores. The M1 Pro that's used in the base 14-inch MacBook Pro models offers either 14 or 16 GPU cores, with 16 being the maximum in this member of the M1 family. For the ultimate in graphics performance, the M1 Max is the chip for you (if you haven't already come to that conclusion based on memory needs), with a choice of 16, 24 or 32 GPU cores.

SSL X-Echo

Delay Plug-in For Mac & Windows

SSL Native X-Echo is a tape-echo plug-in that's inspired by the sound of early hardware units, not least the Echoplex, though it doesn't purport to directly emulate any specific machine. Rather, it gives you the tools you need to tailor your own delay sound. To help capture the vintage vibe, the plug-in makes use of SSL's signature saturation and warmth algorithms plus adjustable wow and flutter but it also includes a newly designed reverb engine for adding diffusion to the delays. It is compatible with all the usual Mac and Windows hosts (VST2/3, AAX and AU formats) and copy protection is via an iLok account, though it doesn't require you to have a physical iLok dongle.

The most desirable characteristics of vintage tape units have been modelled; there's EQ in the feedback loop, soft clipping and the ability to self-oscillate (it is possible to adjust the feedback to over 100 percent) for example. In the name of authenticity, there's also the addition of low-level tape hiss, but while that's always on it's too low to cause problems and does actually make the end result sound more tape-like. A less 'vintage' addition is a Freeze function, which keeps the delays repeating indefinitely. Another is a delay Kill feature, which kills the feed into the delay engine. SSL added these for the benefit of those wanting to create drops or risers in EDM-style compositions. There's also a built-in, variable-depth de-esser to tame sibilance build-up in the delayed sounds. Less esoteric, but no less useful are separate bass and treble EQ controls and the expected wet/dry Mix control. Finally, saturation and wow/flutter parameters are fully variable.

The interface is designed around X-Echo's functionality rather than

resemble a hardware tape delay, and includes a very practical display that shows how the delays from the four 'tape heads' interact. Buttons turn each of the heads on or off. The display can also show the first four seconds of waveform once the DAW is started along with waveforms of the repeats.

Clicking the magnet icon shows note values below the delay tape markers and it is possible to lock repeats to note values using a single mouse click; the tempo may either be set manually, via tap tempo or via host sync. Delays can be sync'ed to the nearest note value, another non-vintage feature, and it's also possible to produce a stereo effect from a mono input via a process that uses a novel phase-shifting technique. A button sets the delays into a left/right ping-pong mode and, with the width control cranked into the over 100-percent zone, the sound becomes very spacious indeed.

"I have to say that it sounds wonderful, and this is due in no small part to the Diffusion control."

So far, I've largely outlined how this plug-in goes about its business but while all that makes it easy to set up, what really matters is how it sounds. I have to say that it sounds wonderful, and this is due in no small part to the Diffusion control, which takes the sound from hard tape repeats at one end to a lovely diffuse reverb at the other. In between are some deliciously floaty ambient treatments that blend delay with a reverb tail, which is attached to each of the repeats. The Saturation control warms things up in a very believable way, while the Wow and Flutter control can go way beyond 'worn transport' emulation to deliver interesting chorus-like effects.

In short, the illusion of a vintage tape echo is very convincing if that's what you're after but there are other plug-ins that can do that. For me, the real value of X-Echo is in how it's able to combine the best of vintage tape echo with modern reverb and DAW-friendly features to create anything from retro slap-back and '60s guitar instrumentals to contemporary vocals and dreamy ambient soundscapes.

Paul White

£ £139 including VAT. Also included in SSL Complete subscription bundle ($14.99 per month).

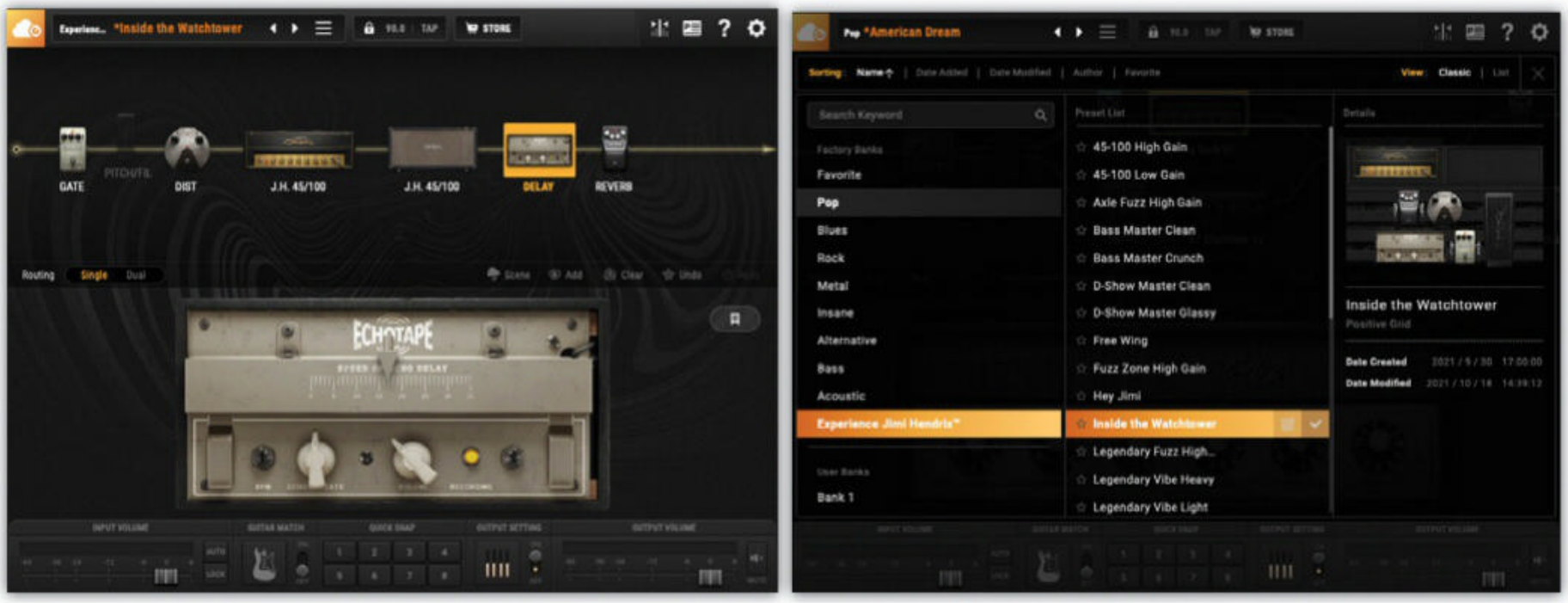
W www.solidstatelogic.com

Positive Grid Experience Jimi Hendrix

Expansion Pack for Bias FX 2 & Spark

When guitar players think of Jimi Hendrix, they associate his unique sound with certain key pieces of audio gear and, of course, his unique playing style. I'm afraid only you can help with the playing-style part of that equation, but when it comes to gear the forums are full of talk about Jimi's use of the Dallas Arbiter Fuzz Face, the Tone Bender, the Roger Mayer Axis Fuzz, the Uni-Vibe, the Echoplex tape delay and the various Marshall, Fender and Sunn amplifiers that he used in the studio.

This expansion pack, which has been approved by the Hendrix estate, requires the Positive Grid's Bias 2 FX plug-in or Spark amp to run and adds the extra pedal and amp models needed to recreate those classic sounds. Owners of the Spark amplifier can buy a version within the Spark app that not only adds the sounds but also shows you the chords for some of Jimi's best-known songs and even selects the right presets for you. There's also the option to explore details of Hendrix's life and career including a timeline and additional info on the gear he used.



One of the benefits of this release being approved by the Hendrix estate it that there's no need to come up with creative preset names — just look up the song you want!

Once you've added the Hendrix collection to your account and downloaded the latest version of Bias FX 2, you'll find the extra bits of gear plus a range of Jimi Hendrix presets that provide the correct combination of amps and pedals, along with the correct settings to emulate specific songs such as the gentle classic 'Little Wing' or the somewhat less gentle 'Purple Haze', complete with Octavia fuzz. Of course you can also combine these new pieces with your existing Bias FX 2 amps and pedals to create your own sounds. There's also a Guitar Match profile as part of the Hendrix expansion that aims to transform the output of your guitar to match one of Jimi's Strats, including his famous hand-painted model. (Thankfully the software doesn't go as far as setting fire to your guitar!)

On careful listening to the original records, you may be surprised to hear that, on some songs, Jimi's guitar sounds rather less dirty than you might imagine — 'Hey Joe' is a good example. However, for the songs where he brings in one of his fuzz boxes, things do get appropriately gnarly and this plug-in captures the raucous nature of those old fuzz pedals particularly well. This expansion pack covers the extremes of Jimi's sound with great authenticity, though it's worth noting that the preset for 'Little Wing' doesn't include the Uni-Vibe that he brought in after the intro; the model is there, but you'd have to add it to the patch yourself if you want that. The various amp setups do a fine job of capturing the essence of the Jimi Hendrix sound, while those vintage fuzz machines really nail the magic and aggression of the low-tech originals. The Echoplex delay also sounds rather lovely, exuding just the right degree of analogue warmth and, of course, there's the classic Uni-Vibe to add that gentle churning phaser effect whenever it's needed.

That just leaves learning to play like Jimi, which is a job for another day. *Paul White*

£ Add-on to Bias FX2 $99.
W www.positivegrid.com

Cherry Audio Stardust 201

Tape Echo Plug-in

I have owned many different tape echo units over the years but the most reliable was my Roland Space Echo, which also delivered cleaner repeats than many earlier tape units. Cherry Audio's inexpensive Stardust 201 plug-in borrows features from the original RE-201 and RE-301 hardware, including the chorus section and spring reverb. Being honest, I was never too impressed by the reverb in the original, as it sounded too twangy and drippy for my own tastes, but it seems that authenticity has been traded for usability in this respect — the reverb implemented here seems altogether better behaved than that in the hardware, and that's a real win in my book!

Stardust 201 Tape Echo

The front panel is visually reminiscent of the original, here sporting a seven-way Mode selector knob with virtual LEDs to determine which of the three replay heads are active. A Mono/Stereo switch, affecting the echo, reverb and chorus, has been added (the original hardware was strictly mono). The Reverb, Chorus and Echo »

» sections have separate On buttons and there are the usual controls for echo Volume, feedback (Intensity) and Bass/Treble EQ, switchable for Echo Only or Echo plus Reverb. The Repeat Rate can be set to Normal or Wide, the latter giving a maximum of two seconds for head three, though there's also a Sync switch that allows for delay times up to eight beats. For those who like the sound of tape slowing down, there's a Motor On/Off switch. There's also a MIDI control section.

Chorus has simple Rate and Depth control and there's an input Volume control with a moving-coil level meter above it to the left of the panel. It is possible to drive the meter into the red to add a little saturation if required. Bottom right is a switch to kill the dry sound, which would normally be used when the effect is set up in an aux send configuration.

Part of the character of a tape echo is down to transport imperfections that add unplanned pitch wobble. Wow and Flutter is emulated here — a single knob controls how much you add — but the designers have also added a separate modulation section, with its own Rate and Depth controls. This goes from subtle to seasick and can work really well in combination with the dedicated Chorus section.

"The reverb implemented here seems altogether better behaved than that in the hardware, and that's a real win in my book!"

Judging the absolute authenticity of the tape delay sound is more difficult, because much depended on how worn the tape was and whether or not you used the recommended tape loops or spliced up your own. But this plug-in definitely captures the overall vibe rather well, and everything seems very well thought-out — for example, the authentic dub-style runaway sound you get when maxing out the Intensity control. I have other emulations of the RE-201 Space echo that cost rather more than this one does — under forensic scrutiny some may score a few points on absolute authenticity in certain areas but, as a musical tool, the Stardust 201 sounds fabulous and delivers everything you could want from an emulation of a tape loop echo. *Paul White*

£ $29 (discounted to $19 when going to press).
W https://cherryaudio.com

Denise Audio Bite Harder Frequency-selective Bit Crusher

If you already checked out our review of the Denise Dragon Fire compressor, you'll be familiar with the Denise 'push-pull' approach to processing. Essentially this means creating an EQ curve to go before the processing, and the option to apply the inverse curve after processing. What this amounts to is that the effect can be concentrated into those parts of the spectrum that need it and be dialled back in other parts. The filters can be set to be subtle or to include dramatic resonant peaks.

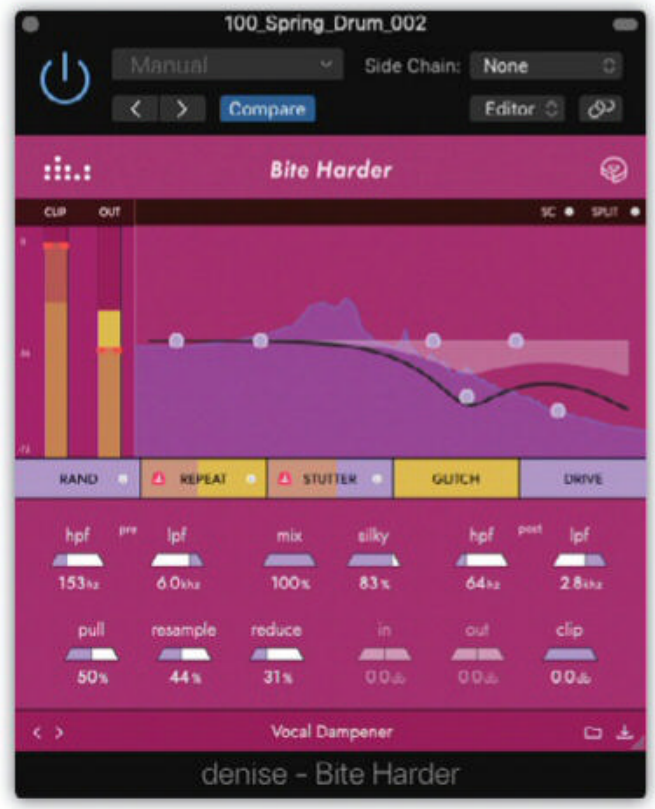
Bite Harder, which supports the usual roster of plug-in formats for Mac and Windows hosts, combines push-pull EQ with familiar bit-crushing and sample-rate reduction functions, though there are further controls to facilitate more creative audio destruction. For example, while the Silky control may be used to smooth out the effect created when adjusting the bit and sample-rate reduction controls, Glitch adds even more nastiness. There's also an effects layer and a drive control.

The controls are arranged as a horizontal row of boxes, four of which act as amount faders plus 12 small horizontal sliders below. In the horizontal row, there's a Random button for generating random EQ curves, plus settings for Repeat (which loops a section of sound) and Stutter (a chopper) that are adjustable in tempo-sync'ed increments. Small, non-latching buttons allow for these two effects to be engaged momentarily via automation. Glitch brings in ringing, non-harmonic components via a ring-modulator while Drive adds some old-school distortion that can get quite spluttery, rather like an early fuzz pedal. There's also a transparent clipper that can get less transparent, in a useful way, if you really push it.

The controls below this govern mix, filtering, levels, bit-reduction and resampling plus there's a Pull control that adjusts how much inverse EQ is applied to the treated signal. The Silky control also resides here.

While these kinds of treatments can work wonders on bland drum loops, more careful adjustment can add useful enhancements to synth sounds, and I managed to cook up a patch that added a bit of reedy-sounding brightness to an EBow part without the end result sounding too obviously distorted. Bass parts are also fair game.

If you are a fan of bit-crushers, then Bite Harder gives you more flexibility than most destruction plug-ins with the push-pull EQ being particularly powerful.



I'm sure it will find applications across multiple genres, though the novel Repeat and Glitch functions will probably appeal most to the EDM fraternity, since they make the creation of stand-out spot effects really simple. This is a serious Plug-in of Mass Destruction, capable of wreaking audio mayhem on a grand scale, so if that's your thing, download the demo version and try it for yourself. *Paul White*

£ £24.47 including VAT.
W www.denise.io

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PODCASTS

Synth Gems 1 - A New Book

Exploring Vintage Synthesizers

Synth Gems 1 is a new book by author Dr Mike Metlay and sets out to document, both visually and in words, over fifty of the most incredible keyboard synthesizers ever made.

From Tape To DAW

Recollections Of The Changing Formats

Paul White and Hugh Robjohns discuss the evolution of recording from the perspective of their own experiences, starting with analogue tape and ending with computer-based digital systems.

Sound From Mars With DPA Microphones

Behind The Brand

René Mørch of DPA Microphones chats to Sam Inglis about DPA's involvement in the NASA Mars 2020 mission and the challenges involved in capturing audio on another planet for the first time.

Public Service Broadcasting

The Making Of Bright Magic

J. Willgoose Esquire, founding member of alternative band Public Service Broadcasting, talks to Caro C about the origins of the band and the surprise success they've enjoyed. He details his recording methods, choice of equipment and the technical challenges of performing live.

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SAMPLE LIBRARIES

Sonixinema
Hang Drum
Kontakt Instrument

★★★★

Sonixinema have something of a boutique vibe as a sample library/virtual instrument producer and their latest release within their 'Experiments' line, Hang Drum, is almost mainstream by comparison. As an instrument, the Hang Drum is a relatively recent development, but will be familiar to many due to its popularity with busking street musicians. As a type of tuned percussion, its roots in the steel drums of the Caribbean are obvious but, in the right hands, the sound can be both beautiful and hypnotic. However, if you have ever fancied picking one up for yourself, you will know that a good example is not cheap and, while it looks deceptively easy to play, it's a skill that takes some mastering.

Sonixinema's virtual Hang Drum may therefore appeal on both price and practicality. It will run in either the full or free version of Kontakt (5.8.1 or above) and contains just shy of 4GB of sample data. The 10 main presets each feature a different performance articulation and a selection of Kontakt Snapshots then customise each sound further using the various effects options built into Hang Drum's UI. That UI contains a number of elements. The Global section provides some macro-style sound-shaping controls, while the Envelope and FX tabs provide further, more detailed, options. The Mix page allows you to blend between four different microphone positions and includes the very neat Proximity macro slider from the Global page as well as options for managing RAM use.

Given how a Hang Drum is often played using hypnotic repeating patterns, perhaps the highlight of the UI is the mini-sequencer/arpeggiator. This includes a pre-stocked preset system and a range of tools for creating and editing your own sequencer patterns (including a randomise function). It's a well-thought-out design that manages to be both easy-to-use and very creative.

The main presets span a good range of articulations including single hits, muted hits, flams, harmonics and a range of more specialised options such as tremolo, pitch bends and side and shoulder hits. The sampling appears to be pretty deep with plenty of round robins and velocity/dynamic layers. Technicalities aside, the sounds are excellent and capture the essence of the instrument very effectively. Perhaps my only disappointment is that there isn't a single, multi-articulation, preset included with keyswitching between the single, muted, flam and harmonic articulations. It would be great to see that added at some stage and even better if the sequencer could also have an extra lane to switch between these articulations.

That said, Hang Drum is a breeze to use, is sensibly priced and sounds delightful. Sales of real Hang Drums to busy media composers may just take a nose-dive. Like many of Sonixinema's products, Hang Drum may be somewhat niche, but it is also very good.

John Walden

£89

www.sonixinema.com

Zero-G
Elements
Kontakt Instrument

★★★★

Zero G's Elements is aimed at those working on creating contemporary soundtracks and sound design, but in reality it would also fit into many musical genres. Many of the samples used in this three-layer instrument were created by recording organic sources, acoustic instruments, and particular environments, which include wind, rain and water, though there are also some synth-like sounds. The full version of Kontakt v6.6.1 or later is required.

Elements is described as a Wavetable Hybrid Synth and comes with a Wave ROM of over 23GB, the presets divided into categories for easy access.

The instruments include a raw-sounding old English Cello, Classical Guitar, Electric Guitar, Felt Piano, Hardware Modular Synths, Viking Horns, Mallet Instruments, Metal objects, ethnic percussion, drum loops and so on.

On opening Kontakt you'll find normal and Lite versions as well as a version that allows audio samples to be imported into the third layer via drag and drop. I found that the normal version really hammered the CPU on my 12-core, 2012 Mac Pro, sometimes pushing it over the edge, even with a 256 buffer size (one of the more powerful i-series processors is recommended), but the Lite version behaved perfectly with no apparent loss in sound quality. However, I did find that just a few of the samples didn't play at all in the Lite version. The manual informs that the Lite version has a less CPU-intensive engine, uses less memory and drops some features, but I don't think anybody would feel short-changed if they had to use it.

Some of the 500 or so factory sounds are playable as conventional instruments while others include note transitions, usually fairly simple ones. Others include velocity switching, plus lots of examples arranged as arpeggios or pulses, the tempo-sync'ed sounds optimised for use in the 75 to 110 bpm range. The native Indian flutes are particularly evocative, as

are the classical guitars, and the vintage cellos are fabulous, but there's so much to explore here, including some unusual choral sounds. The library really is huge, based on over 8500 individual samples.

While Elements may not provide everything you need to tackle any scoring situation, if you already have some good orchestral sounds then you should be able to find something here that works. The breadth of sounds on offer is impressive so your only challenge is to find time to explore them all. *Paul White*

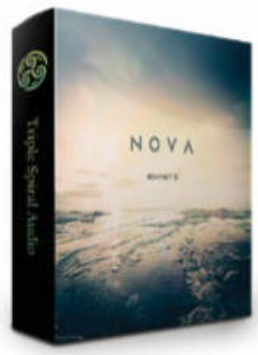
£64.01

www.timespace.com

Triple Spiral Audio Nova

Kontakt Instrument

★★★★★



Triple Spiral Audio's Nova is a sample-based virtual instrument designed for the full version of Kontakt 6.5.1 (or later). As a collection of textures, pads and soundscapes, like many of the other titles within the TSA catalogue, it seems to be primarily aimed at the media composer. Like the majority of TSA's products, however, it is modestly priced so, even if you aren't yet scoring Hollywood blockbusters, it is very accessible.

Nova is built around 130+ sound-sources and totals 1.8GB of sample data. These are accessible via a dedicated Kontakt front-end that allows you to blend two of the core sound-sources to create your final sound. While this is not a new approach, and the features within the sound engine perhaps don't break too much new ground, it does provide plenty of flexibility. Although the focus is very much on the atmospheric and evolving sounds, you do also get a second front-end with a similar UI that provides a range of percussion/drum sounds. Potentially, therefore, you could write a full 'ambience' musical cue using just the Nova and Nova Percussion sound sets.

Both engines include a set of presets supplied as Kontakt Snapshots. These are well organised and, for the pads and textures, do a great job of demonstrating what the library is good at. There are some excellent ambiences, and the Motion, Pad and Soundscape categories are full of excellent extended textures that have plenty of sonic motion. The two layers each offer ADSR and a filter with its own envelope. However, there is also an arpeggiator and the option to control the volume of each layer independently via its own LFO; if you want to create more rhythmic sounds or pulse effects, that's perfectly possible. Add in a range of effects including reverb, delay, modulation, distortion and stereo imaging, plus a neat Random button that keeps all your settings intact while simply randomising the two sound sources used, and you are clearly not short of options. Oh, and the icing on the cake is that you can also drag and drop your own samples into a special User Layer in each of the two sound slots and then manipulate those with Nova's sound-design features; very cool. The Nova Percussion engine doesn't support drag and drop but the sounds are suitably cinematic and very usable.

Nova might not have a Hollywood blockbuster price tag, but the sounds are excellent and the UI is both easy and flexible. Sonically, it is most certainly a niche library, but if you need to add some additional ambience, pad or soundscape-style options to your sonic arsenal, Triple Audio Spiral's Nova is a bit of a bargain. *John Walden*

€39.62

www.triplespiralaudio.com

Spitfire Audio Albion Solstice

Kontakt Instrument

★★★★★

Billing itself as a "celebration of un-notated traditions", Spitfire's Albion Solstice may be the most surprising addition to the iconic series. While it may at first seem, both from the billing and a jaunt through the product, that Solstice is a collection of "all the remarkably cool things we couldn't find a way to fit into the other packages", there is a unifying sensibility to the hefty 73GB collection. And while clearly eclectic amongst its Albion peers, Solstice nevertheless fits aesthetically into the series as a whole, while providing an earthy and folkish set of options to lovers of the Spitfire sound.

Beginning in the main Solstice Orchestra collection, we encounter folders of strings, brasswind combos, folk pipes and flutes, several collections of percussion instruments, a 'band' folder, and a choir. The surprisingly substantial string family is divided into folders of sextet, octet and bass options. In most of the instrument folders there are options for standard techniques, an extended techniques instrument, as well as what Spitfire calls 'evos' and 'motors'. Evos are organic pad-like instruments based around individual or ensemble techniques, and add a substantial portfolio of such options for the composer. The motors option provides various repeated patterns recorded at 90 and 110 bpm, with sliders to adjust tempo speeds from this starting point. Motors may seem limited and disappointing at first, but yield a great deal of usefulness upon further exploration. While standard repeating note patterns on strings may be useful, the quirky repeating 'na nas' and expressive layering options in the choirs folder are a unique touch in the library, while the strummed chords options in the Gut Circle (acoustic plucked instrument) folders provide great track starting points and backgrounds. One practical use with motors is layering them with other similar pattern instruments to create greater depth and realism to the sound.

Moving through the instruments, the brass and winds combo folder is great for ensemble layering, while the pipes folder contains a nicely sampled pan pipe. A particular favourite was the Generator Trio, including a beautifully sampled and highly playable expression and gentle electric guitar. The choir folder also provided a large sampling of folk-like techniques. Also included are a substantive collection of options under the Cassette Orchestra (utilising Spitfire's eDNA engine) and the Drone grid (familiar to users of LCO Textures).

Whether you are looking for great layering options or want to create an orchestral mix using instruments which are just a bit different and folk-like, Solstice opens up into a surprising array of choices capable of adding depth to many a mix, creating a package whose content will take many hours to fully explore. In the end, Solstice is a package quite capable of greatly expanding your sonic palette. *Mark Nowakowski*

£399

www.spitfireaudio.com

Audio examples of this month's libraries are available at www.soundonsound.com.

Q Can I combine mic polar patterns for greater low-frequency extension?

Is it common to combine an omni mic for LF extension with one or more directional mics for higher up? I've not given it much thought before, though as I've seen something like that used in hardware recorders. Should I assume there are already established arrays for this sort of thing (ie. the inverse of a two-way speaker, with a crossover separating the omni LF pickup from the more directional mids/HF of the other mic)? I realise directional polar patterns get more omni at lower frequencies, but I was thinking rather of the greater frequency extension of omnis.
Matty Jones via email

Technical Editor Hugh Robjohns

The short answer is yes, it is common. The longer answer is there are lots of ways of doing this, each with their own pros and cons. It is very common in classical recording circles, for example, to add 'omni outriggers' (spaced omni capsules) to the main coincident or near-coincident stereo array. In general, the outriggers bring two useful gifts to the party: they add a spaciousness (particularly at low frequencies) that is often missing from coincident and near-coincident arrays, which gives a better sense of space and depth to the recording. And secondly, they support the bottom octaves below 80Hz, where directional mics typically start rolling off and become progressively weak. Ask 10 recording engineers how to rig this kind of dual configuration, though, and you'll get 11 different answers!

The easy option is to rig an ORTF pair as the main array, with spaced omnis on the same plane but about 60-90 cm (2-3 feet) apart. However, since omnis inherently pick up more room sound the perspectives of the two arrays is quite different, and often I've found it better to rig the omnis separately around 2-3 feet closer to the source, in front of the ORTF array, particularly with smaller ensembles. Some experimentation will be required with the actual separating distance and omni mic spacing to get a suitable match of stereo recording angle to avoid badly mismatched stereo imaging from the two arrays. In situations with large ensembles, or pipe organs, it might prove better to place the omnis well behind the main (directional pair) stereo array, to emphasise the spaciousness and room ambience.

In this kind of application, though, there is no requirement at all to introduce any specific filtering to 'cross over' between the two arrays; it's just a question of panning the mics in each array hard left/right and mixing them together with an appropriate balance to create the required sound. In general, though, either the spaced omnis or the directional (ORTF, etc) pair will dominate the mix, with the other array supplementing at a lower level. If the two arrays contribute similar levels there is a risk of comb-filtering effects.

However, where time and technology allows you could also introduce low- and high-pass filtering to 'cross over' between spaced omni and coincident mics (or some other stereo array), respectively, if you really wanted — and it's something Blumlein experimented with in the 1930s. The idea would be to maximise the inter-channel timing differences at low frequencies and inter-channel amplitude differences at higher frequencies, on the basis that these are separate mechanisms employed by the human hearing system. A crossover frequency around 600-700 Hz is usually preferred, but some juggling is required of the physical mic configurations and crossover frequency to match stereo recording angles between the two arrays across the full bandwidth, so that the stereo width is stable at all frequencies.

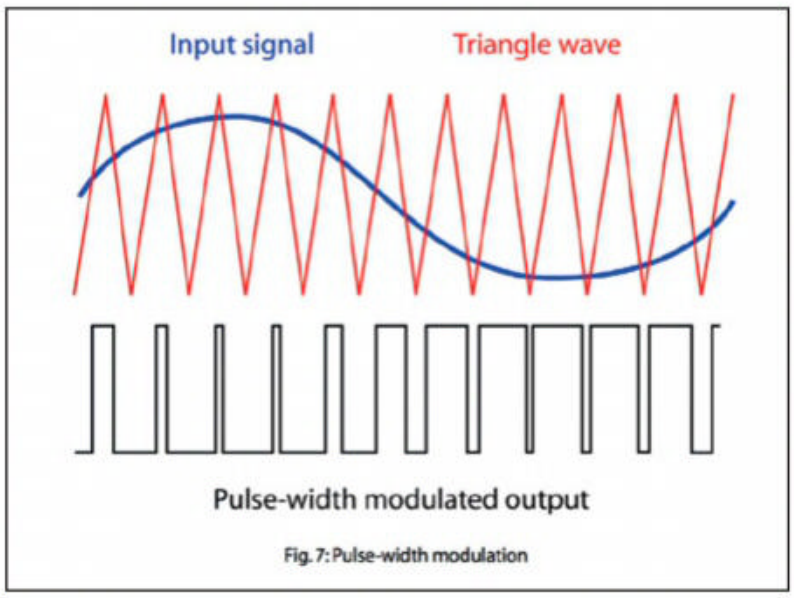
In the Tresmic array employed by this Olympus LS-14 handheld recorder, a central omni mic is added to a directional L-R stereo pair to extend the frequency response.

Q Is your Class-D amplification article inaccurate?

I'm a long-running technical electronics repair and design engineer working in music for film. In David Mellor's article What Is Class-D Amplification? [*SOS* June 2006: http://sosm.ag/d-class-amplification], at figure 7 I think he may have the waveform representation (colour blue) reversed in comparison to the PCM signal presented; if more square-wave duty cycle pulses are high than low at a given time, the analogue waveform would also go high (polarity-wise). I know it's a petty thing but it might confuse some people. Also, at the end, he says in a class D you can't find transistors because of large transformers, but this is actually the reverse comparatively to conventional amps.
Chaz Labrecque, via email

Hugh Robjohns

Hello Chaz, and thanks for getting in touch. I can see where you're coming from, but I think you're seeing the blue waveform as the (filtered) output of the PCM signal, which is not what the text describes — I suspect your experience in the topic might be causing you to read and see what you know, rather than what's actually there and intended for novices. David is explaining how a PCM signal can be produced from the output of a simple comparator receiving the (blue) input signal and the (red) triangle wave.



The 'figure 7' diagram from our 2006 article on the different classes of amplification, as discussed in the main text.

What you say about the analogue waveform going high is true, but it's not what David is trying to illustrate. To regenerate the same polarity of (input) waveform from the PCM signal as shown, it would be necessary to invert it, but that's a complication that David has overlooked for simplicity in his explanation.

Regarding your point about transistors, I think what he's trying to say is that the Class-D amplifier circuitry is typically far more compact than a traditional Class-AB design, and that its linear power supply and transformer might well be the largest physical section of the overall design (not that many Class-D amps now use linear power supplies, of course).

JULIAN RODGERS

The latest Pro Tools and Eucon updates offer much to both pro and project-studio users.

In the closing days of October, Avid announced the release of Pro Tools 2021.10 and Eucon 2021.10. Although not a milestone like 2021.6, which offered huge performance gains courtesy of the Hybrid Engine, this double release brought some useful improvements for music production. There's also some very welcome news for HDX users anticipating upgrading their Apple computers to one of the new M1 Pro and M1 Max machines announced in October.

Going Native

The word 'native' has always been one that needs careful attention in the Pro Tools world. The potential for confusion about whether a 'native' Pro Tools system is software-only Pro Tools or the confusingly titled HD Native hardware can get messy, and the same has to be said of the language used to describe Apple Silicon compatibility. At the time of writing, Pro Tools software is compatible with Apple Silicon-based Macs via Rosetta 2, but it is not yet natively supported. However, users of Pro Tools HDX systems haven't been able to use their hardware on M1 Macs because although applications can run using Rosetta, drivers cannot. Without a suitable HD driver, HDX systems have been excluded from the 'M1 via Rosetta' party.

This has changed in 2021.10 with the release of a Universal Binary HD driver that supports Apple Silicon natively. This means that hardware-based Pro Tools systems can now run on M1 Macs. Only when a fully M1-compatible version of Pro Tools software is released will the program be able to run completely natively on Apple Silicon, but until then, HDX users will be able to use their systems, with an appropriate chassis, on an M1 Mac.

Remote Control

The release of Carbon was proof, if it were needed, that Avid haven't forgotten about project-studio users. Those who have used it — and I've had a unit since it was released — will know just how well the features and workflow address the needs of the user who needs a latency-free Pro Tools tracking experience for recording bands. This kind of user is typically working without the benefit of assistants, so the remote and recallable mic preamp settings introduced to Carbon in 2021.10 will be extremely welcome.

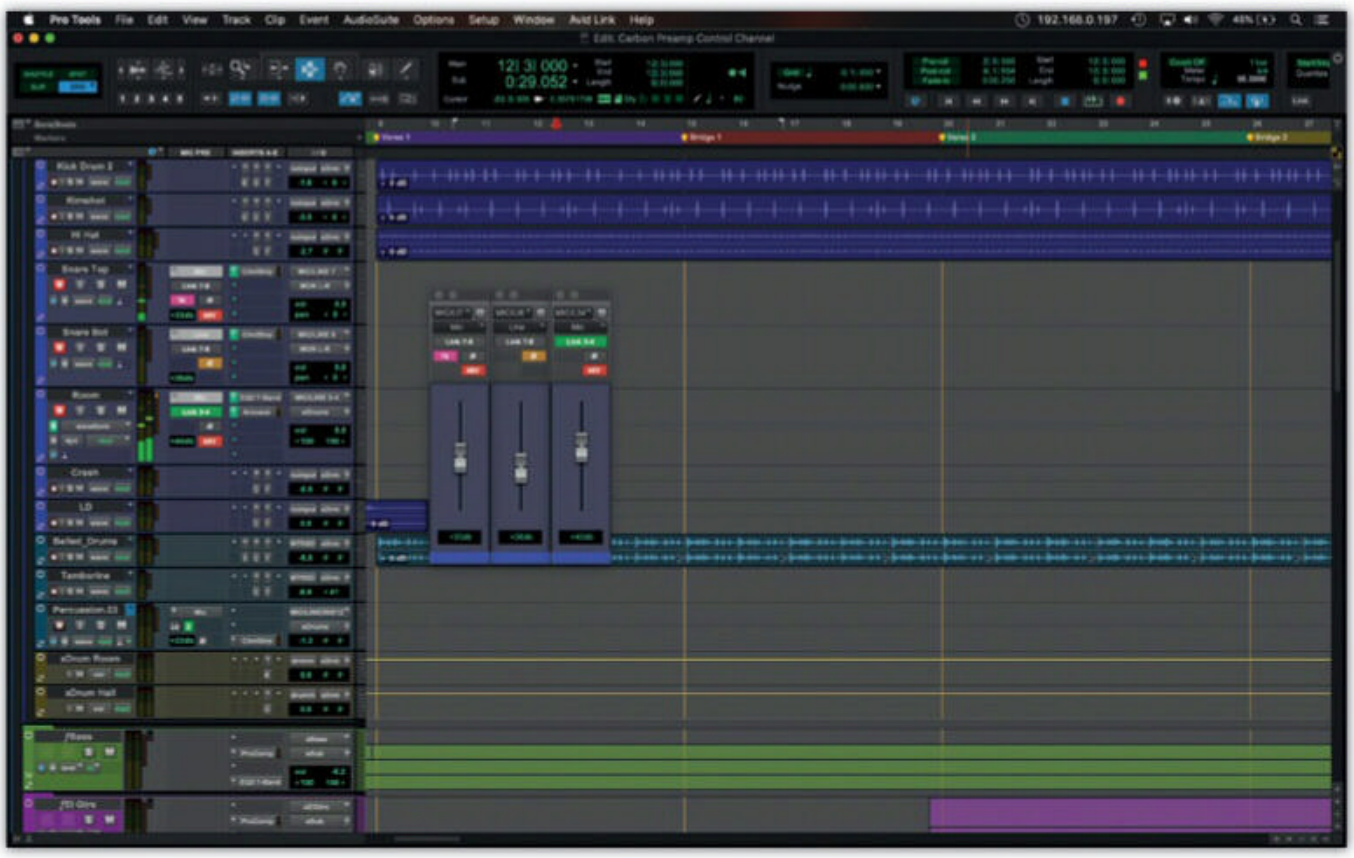
The eight built-in mic preamps in Carbon can now be remotely controlled from Pro Tools software or the Avid Control app. The ability to tweak gain, polarity and phantom power from an iPad while setting mics is something which will make setting up sessions much easier, and with the ability to remotely control preamp settings comes the possibility to save and recall preamp settings with a Pro Tools session or template.

Recall of preamp settings can of course be done from the Import Session Data dialogue, but an extremely useful extra feature for busy tracking sessions is found in the Hardware Setup window, where you can opt to isolate the preamp settings so they persist when changing sessions. Mic preamp settings can also be stored as part of a track preset, so specific recording setups can be recalled quickly. Track presets also feature in an update to the Workspace browser, with a new menu item to access track presets in the browser.

The preamp controls can be accessed from a new section of the Mix or Edit Window track header, and a floating window can be detached and placed in the UI in a similar way to floating send faders. The use of colour is a nice feature of the Carbon unit: it uses multicolour illumination of buttons and meters to guide the user through the current settings, and the way this colour is carried through into the software UI is helpful, showing the same colours for different impedance settings on both the hardware and software.

Unfortunately at present there is no way to link or daisy-chain Carbon units, so in a fully expanded system with 16 additional preamps connected via ADAT, only the eight preamps in the Carbon would be remotely controllable. Remote control of third-party preamps is possible but the choices are limited at present. A dedicated Carbon expander mic preamp unit would be a lovely thing but I won't hold my breath, particularly considering how personal mic preamp choice is.

On the subject of the ADAT ports in Carbon, these were previously ADAT-only, but with Pro Tools 2021.10 they can be individually switched to S/PDIF at sample rates up to 96KHz — useful for those who need it.



The preamps on Avid's Carbon interface can now be remotely controlled directly from within Pro Tools, or via the Avid Control app.

Komplete Kontrol Integration

Pro Tools 2021.10 includes integration with Native Instruments' Komplete Kontrol system, which brings together

The Retain Mic Pre Settings option in the Hardware Setup window allows you to switch projects while keeping your Carbon preamp settings the same.

the vast content libraries available from NI with their very popular MIDI controller keyboards. The micro-sized M32, and the full sized A and S series, all use a system of deep integration with Native Instruments products allowing content browsing and control of patches as well as a level of DAW control such as basic mixing, session navigation and transport control. It's a long way from the kind of control offered via Eucon but it is a useful extension to what is available using MIDI or HUI.

This additional control is provided courtesy of a new MIDI Controller Profile, available from the MIDI Controllers tab of the Peripherals window. The new Komplete Kontrol profile joins the M Audio keyboard profile, which also offers additional parameters.

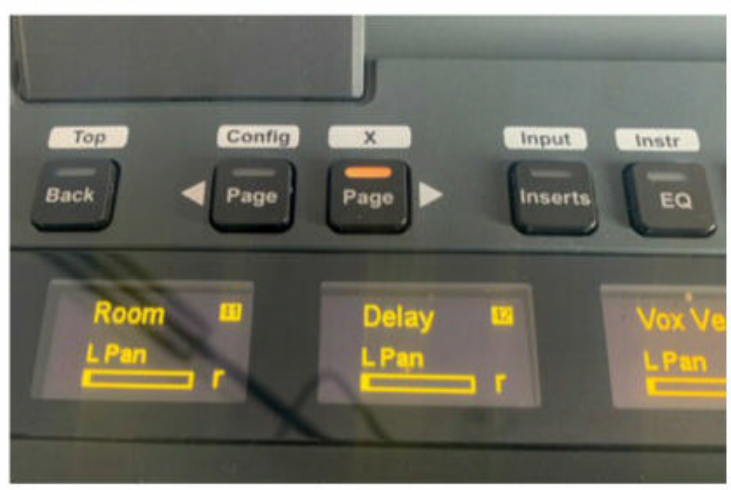
Flexible Track Routing

Multichannel tracks used to be principally of concern to those working in post-production, but the recent developments in Dolby Atmos, with Apple introducing Spatial Audio to the Apple Music platform and integrating Atmos into the recent Logic 10.7 release, provide further evidence that music production is moving beyond straightforward stereo. While post mixers have been working in surround formats for years, for the majority of music mixers, the jump from stereo to Dolby Atmos is a large one. Managing routing to and from busses of different formats has always required some manual intervention, for example with Downmixer plug-ins being used for fold-down operations and use of sub-paths.

Pro Tools 2021.10 introduces flexible track routing, which means Pro Tools can now route to outputs and sends of any format without manual intervention. This downmix routing intelligence means that the user can route virtually, without restriction, to outputs and sends of any format. There are exceptions, though: specifically, it can't be used with Inputs, Objects or hardware inserts.

Eucon 2021.10

The new update to Eucon includes swap layers, which are a feature available on the big Eucon surfaces like the S6. There are also performance improvements, which include changes in the way data is exchanged between the surface and the host computer, to alleviate the lag which can happen when large surfaces are being used with very large sessions. These changes affect all Eucon surfaces, but the bigger the surface, the greater the difference they will make. Users of dual-operator S6 surfaces have reported very significant improvements, but someone with an S1 running modestly sized sessions is unlikely to see much change.



One of the changes to Eucon is that the previously unassigned 'X' key now defaults to acting as a Bypass button.

Something which is going to make a big difference, though, is the introduction of a dedicated Bypass key to the S1. The 'X' key was previously identified as being currently unused but with 2021.10 it is available as a bypass for inserts: a small but very significant feature. There are other improvements, such as the Avid Dock's monitor knob, which is very useful to users without a dedicated monitor controller, as it can be mapped to a master fader in Pro Tools: do this by right-clicking on the master fader's track name and select Eucon Monitor. The monitor knob is now touch-sensitive, so you can see the current volume just by touching the knob. The resolution of faders is now higher below -10dB (I hadn't noticed that they lacked resolution below -10dB, but more is more!). New Softkeys have been added to the already comprehensive list already available: specifically, I/O, EQ Curve, Object and Meters and Faders.

Pro Tools 2021.10 suggests that the pieces are coming together for a native Apple Silicon release before too long. While we don't know when that will be, the introduction of the new M1 Pro and M1 Max, the Pro Tools Hybrid engine as released with 2021.6 and 2021.10's introduction of the Universal Binary HD driver mean that, particularly for HDX users with Macs, the future looks good.

High Contrast

Pro Tools 2021.6 updated the Colour Palette to include a slew of UI customisation options. This has been updated in Pro Tools 2021.10 to include a High Contrast UI scheme which is aimed specifically at making Pro Tools more accessible to users with a visual impairment. This update includes settings like the global colour control of text and lines, and the ability to put white borders around text fields.

From plucks to pads, Cubase's Sampler Track makes it easy to design your own sounds.

JOHN WALDEN

In last month's workshop, we considered how you can combine some of Cubase's stock plug-ins to apply some creative sound design to a 'live' audio source. In this second instalment focused on designing your own sounds, we will take a different approach: building new 'playable' sounds from a single sample. To give our discussion a focus, I'll take a couple of sound-design classics — a short (non-sustained) lead sound and a sustained pad-style sound — and use the Sampler Track, which is available to all Pro, Artist and Elements users, to create them. You can audition the sounds in progress, courtesy of the audio examples available on the *SOS* website.

While the Sampler Track may not be as powerful or sophisticated as, say, HALion or Kontakt, it certainly has enough options to make DIY sound design interesting. Importantly, it's also very easy to use, because fundamental to the Sampler Track's approach is that you work with a single sample. You can put almost any sample to good use for sound design but, for this demonstration, I recorded a single sustained note from an electric guitar into my project, and used this as the starting point. Having selected a sample such as this, a quick drag and drop into the Sampler Control panel (in the Project window's Lower Zone) will add a Sampler Track to your project. Then you're ready to get tweaking.

Short & Sweet

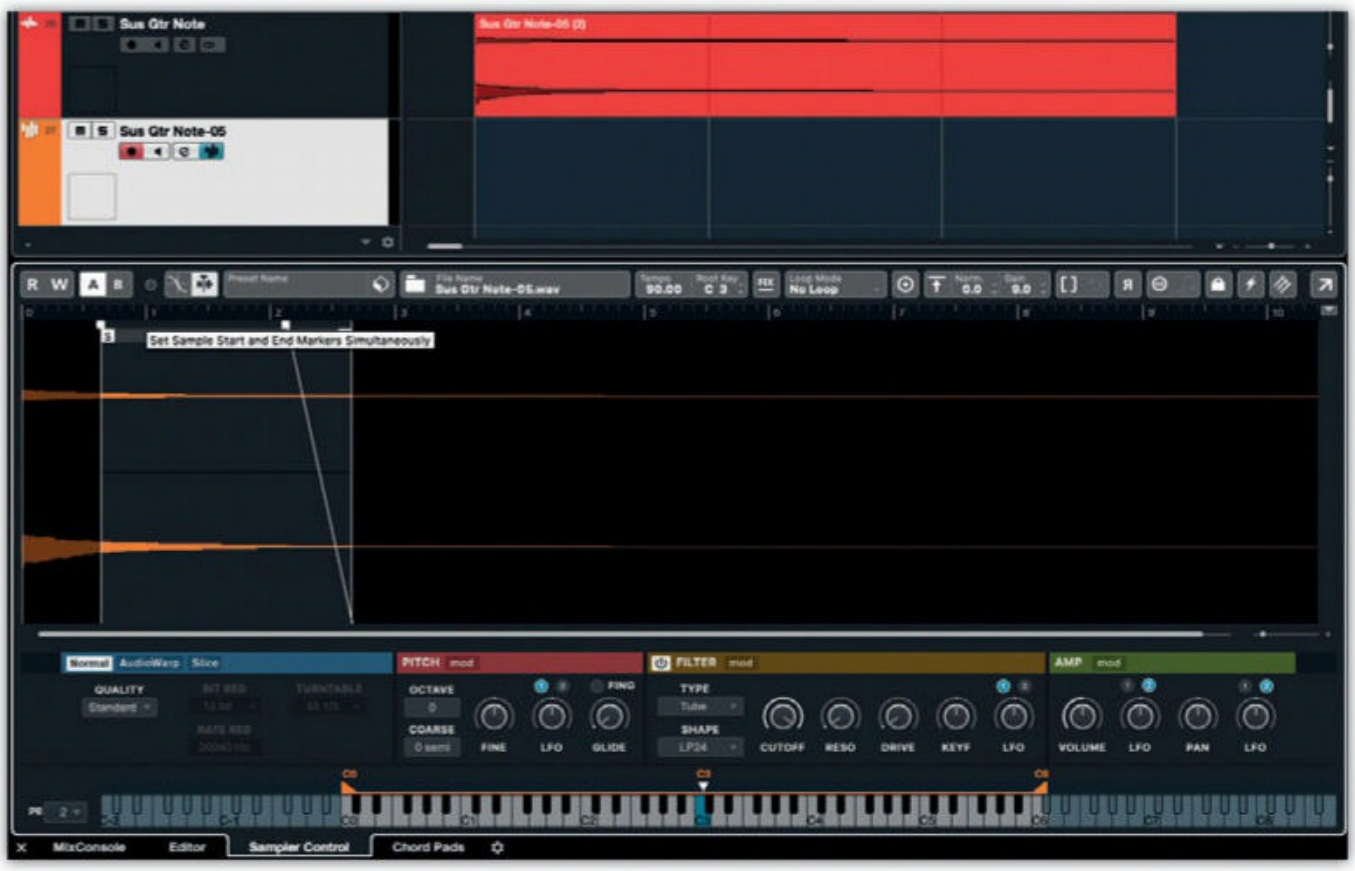
Starting with all settings at their default values (assume I've left them that way if I don't mention a specific setting below), the sensible first step is to start in the Sampler Control's waveform display, and define the portion of the sample you wish to use. This provides Set Sample Start and Set Sample End locators (both with fade options if required) that you can drag into position. A good starting point is to leave the Start locator at the beginning of the sample and adjust the End locator to get something close to the length of the sound you're trying to create. Usefully, if you then hover the mouse between the two locators, a grey bar will appear that connects them, and you can then slide both along the timeline, preserving the selected length while auditioning other sections of the sample. In this case, I ended up with a selection just under a second long but avoiding the very start (where there was some unwanted pick noise), with no fade-in and a short fade-out.

The next sensible step involves making decisions about two settings: Loop Mode and Playback Mode. As the aim was a short sound here, I used the Toolbar to select the 'No Loop' Loop Mode, in which the sample simply plays once whenever it is triggered. Second, in the Playback Mode (blue) sub-panel, you must choose between Normal and AudioWarp modes; the other mode, Slice, is designed for working with drum loops. Normal mode is old-school sampling, whereby playback of the sample at different pitches is achieved by speeding up/slowing down the playback speed. Steinberg have included some cool choices, so don't dismiss this mode until you've tried it, but in this case I went with AudioWarp mode, in which the Sampler Track engine does some time-stretching and formant preservation as well as pitch-shifting when playing back the original sample across the MIDI note range.

Next, it's a good idea to finesse the amplitude of the sample's playback using the Amp panel. If you click on the Mod (modulation) switch in the green panel header, the waveform display changes to show a fully editable amplitude envelope. In this case, all I've done is create a smoother volume decay over a short time period and set the Sustain node to zero (so there is no sustain volume if a MIDI note is held). Top left of the Amp Mod panel, I've selected One Shot Mode (the amplitude curve will not cycle when a MIDI note is held longer than the defined amplitude curve) and Velocity to 100 percent, so that the amplitude modulation is very sensitive to incoming MIDI note velocity.

As you can hear in the audio examples, with the addition of just a little reverb and delay this is already a perfectly playable lead/melody sound. However, our next stop is the Filter panel, with which we can modify the tonal character of the underlying sample. I've selected the Classic filter type and a 12dB low-pass filter, with the Cutoff centred around 160Hz and a high Resonance value (70 percent). Via the Filter's mod panel, I've then created an envelope to modulate the Cutoff value. The AMT slider (far left) can be used to control how strongly the modulation operates, while the Velocity setting (top left) is set to 75 percent and this allows you to also control the strength of the filter modulation via your MIDI performance.

The final sub-panel, Pitch, offers similar modulation



Any sample can be dragged and dropped into the Sampler Control panel to create a new Sampler Track.

The filter modulation envelope was used to add a little tonal character to our lead sound.

options for the note pitch. You can get 'experimental' here but for our playable lead sound I simply added a very small, rapid, pitch 'wobble' over the attack portion of the sound and, again, I added some MIDI velocity sensitivity to this modulation. The overall effect is modest, but it just adds a little extra character.

As is hopefully demonstrated in the audio examples, we now have a very playable DIY lead sound, created in just a few minutes from one of the most boring source samples you might imagine. And if you then start adding in some further effects — reverb, delay, distortion, modulation, etc — you can easily up the ear-candy levels even further.

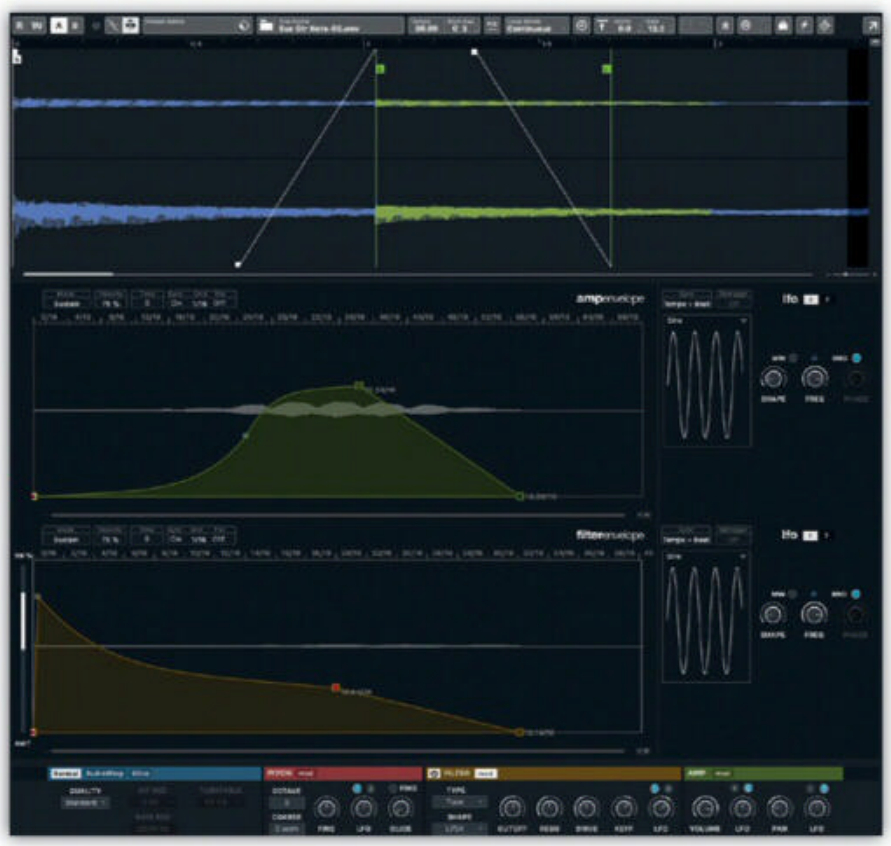
Extra Padding

If designing a pad sound from our source sample, we can follow almost exactly the same workflow. The key difference, and the trickiest bit, is creating a section within the sample for looping when a note is held for an extended time. In the main Toolbar, I've enabled the Snap To Zero Crossing button and selected the 'Continuous' Loop Mode. The latter adds left and right loop locators (in green) to the main waveform display, while the former ensures that when you position these, they snap to points least likely to result in audio glitching.

The trick is to define a loop range that cycles as smoothly as possible. Again, you can do the 'hover and drag' manoeuvre to move both locators simultaneously while auditioning. You can also add a fade-in/fade-out to help smooth things out and there are alternative Loop Modes to explore. There can be quite a bit of back-and-forth experimentation required here and getting a perfectly smooth result may simply not be possible with some source samples.

In our example, I ended up with a loop range of just over a second, which starts just over a second into my sample. When sustaining a note longer than one second, you could hear the 'looping' kick in but, in this particular case, I thought it actually added to the sound by creating a gentle pulse effect.

In the Playback Mode (blue) panel, I opted for Normal mode, since when notes were sustained I felt AudioWarp produced some undesirable artefacts. Normal mode also meant that the 'pulsing' created by my loop region changed speed with the pitch of the triggering note, and I felt that this added nice some extra movement to the sound rather than being a problem.



A composite image showing the loop region, amp envelope (with slow attack) and filter envelope (with LFO modulation) used for our pad-style sound.

As shown in the screenshot, I created extended Amp and Filter envelopes, providing both a gentle fade-in and a gradual tonal change as a note was held (adding a sense of movement). I also made use of the Sampler Track's other modulation option: the LFOs. The two independent LFOs can be used to modulate pitch, filter cutoff, volume and pan. You get a choice of LFO waveforms, the option to change the waveform shape, add Mod Wheel control, and sync the LFO speed to your project tempo. In this case, I simply used LFO1 to gently modulate the filter cutoff, adding a further tonal 'pulse' to each sustained note. Job done!

Lock/Replace

Once you've been through the Sampler Control workflow a few times, DIY sound design is an accessible process. I'll leave a detailed discussion of all the controls and settings for another time but, as a final pointer, one is worth noting: the Toolbar's Lock Parameter Settings button (padlock icon). As the name suggests, if you engage this, all the Sampler Control settings for the current Sampler Track get locked, helping avoid accidental changes to your carefully crafted sound. A very useful exception to this is that you can still change the underlying sample. So if you've created settings for a lead sound (for example) and want to explore how they might work applied to a different sample, simply drag and drop a new sample in and see what happens. I've included some audio examples of this process so you can hear how this might work. It's a easy way to create new sounds and, once you find something that's interesting, you can always unlock the settings and fine-tune to taste.

ROBIN VINCENT

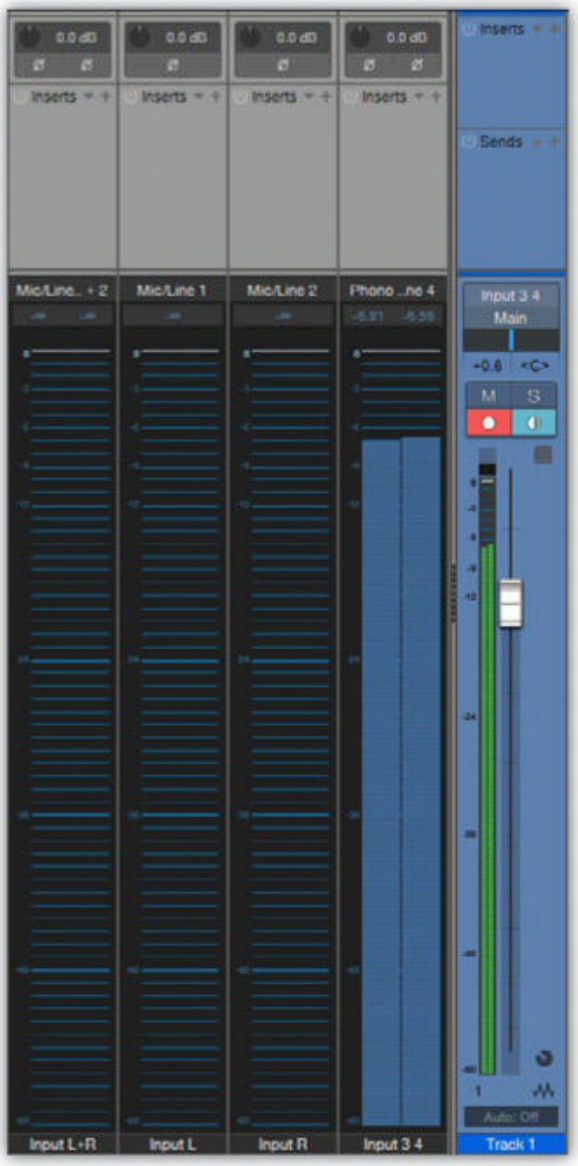
We talk you through the various metering options in Studio One.

After our ears, meters are probably the most important tool in the sound engineer's toolbox (that and a good flask of tea). The meters in the Studio One Console are nicely designed, ever present and very useful. Sometimes we glance at them briefly but with a bit more focus we can use them to improve our mixing game.

If you open the Console you'll find the faders tend to be quite small and humble, but you don't have to put up with that. As you pull out the top of the Console, the inserts space stretches to accommodate the new space. But if you click that line above the fader section and beneath the sends you'll find that you can stretch those faders nice and tall to give you a more satisfying view. The taller the faders, the higher the resolution you have to visually monitor the meters and control your levels.

Faders

The fader section consists of the fader with the metering to the left. You'd expect there to be a peak light, which lights red to congratulate you for reaching the top, directly above the meter. However, there's no peak light on playback channels in Studio One even though it looks like there are; only the input channels and main outputs have peak indicators.



Dragging the Console window up not only makes the faders longer, but it increases the height of the meters too.

Above the Mute and Solo buttons, and to the left of the pan value, is a number that shows the current position of the fader. You can enter a value and the fader will jump to it if you have some precise settings in mind. There are a number of metering options available in the Console. Right-click on the actual meter to display the menu. There are two different metering mode menus depending on what you are right-clicking. For the regular Console channels the options will be Peak, Peak/RMS, Peak Hold, Hold Length (peaks are being held and then fall with a degrade speed defined here), and Pre-fader Metering. Output channels also include the K Metering System, which I'll get onto in a moment.

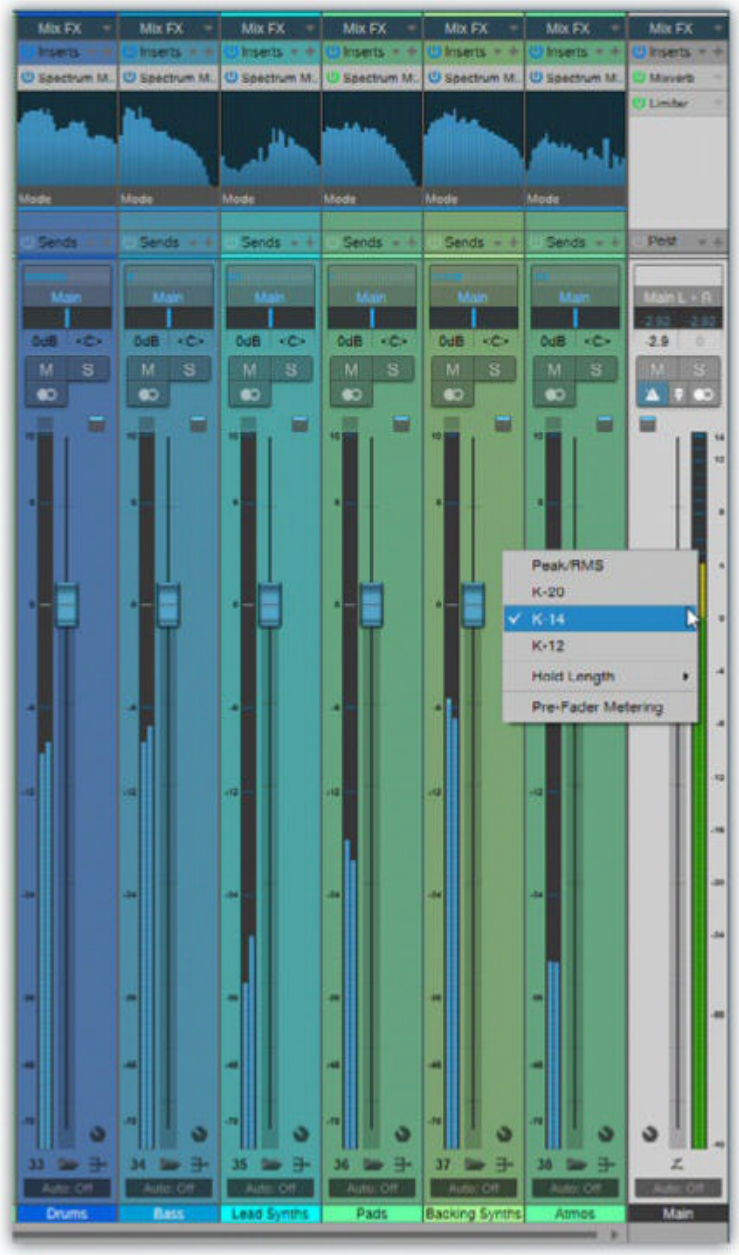
Peak meters continuously measure the audio level at a very fast resolution and translate the highest value at that moment to the meters. The **Peak/RMS** option combines the peak meter with RMS levels. The two slowly moving white lines show the RMS or 'Root Mean Squared' value, which is a kind of average loudness of the total waveform, which responds in a similar way to the human ear. You can enable **Peak Hold** to have the peak level leave behind a little indicator of where it was for a certain length of time, so that you have more time to read the meters that are changing very fast. You can specify how long these peaks hold for or how quickly they get updated. If you want to see what the level is like without the change the fader has applied, then check the **Pre-fader Metering** button.

It should be noted that all of these settings are global. It may feel like you're only changing it on the channel you've just right-clicked on but you are changing it for every channel.

K Metering

Studio One supports the K metering system on the output channels. K metering was designed by sound engineering legend Bob Katz as a way of dealing with the variation in loudness of different source material. Ideally it should be used in conjunction with building a monitoring system around it, but simply using the metering system on its own could have a major impact on your mixing.

Whenever we are mixing there's an inclination to be drawn towards the point of zero. When we see VU meters our analogue brains find themselves looking at hitting that 0VU level — it's just human nature. In analogue systems that's fine, and if you go a little over then that's still fine. In digital systems, like Studio One and other DAWs, anything over 0dBFS is a digital clipping nightmare. We know that we should stay well away from that point of distortion but we are still drawn »



Output channels have additional metering options, including the K metering system.

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» to get the 'most' from our meters. The K metering system can help with this.

What the K metering does is pull that 'zero' point down to give you a more useful and less dangerous point to aim for. Nothing has actually changed except for the numbers at the side of the meters. The actual 0dBFS point is still in the same place, but we've labelled it either +20, +14 or +12, depending on which meter you've selected. These settings put your 'target' zero at -20, -14 or -12 dBFS. It's a genius bit of misdirection. We can now happily aim for that zero point on the meter while retaining an appropriate amount of headroom.

The idea was that for highly dynamic material like orchestras and ensembles then you use a K-20 meter. For pop and rock you'd use K-14, and for melt-your-face electronic music you'd use K-12. When you select a K metering option Studio One changes the colour and speed of the meters to better reflect a more traditional feel.

So, if 0dBFS is the danger point in a digital audio workstation, then why does the playback metering in Studio One go up to +10dBFS? Studio One is not particularly interested in how hard you push your individual tracks, which might explain the lack of peak indicator light. You have an extra 10dB of gain in the fader, that you can use to bring up lower-level recordings and balance your music. What is important is that at the mix bus the sum of the tracks remains below zero — that's where it counts.

Input Monitoring

So far we've been talking about the metering on the output and track Console channels, so let's turn our attention to input monitoring. If you enable the inputs in the Console you'll see that Studio One gives us nice fat input meters and does away with the fader all together. While you can monitor the metering through the audio track's channel strip, the meters on the input channel are far more awesome. First of all, these are large enough to be seen from the other side of the room where your microphones, guitars and other instruments probably are. And secondly, they emphasise the last 12dBFS before the zero point. Those thick lines on -12, -9, -6 and -3 dB are warnings that you really shouldn't be messing about in this space. Peaks of around -12dBFS are probably what you want to be aiming for in the vast majority of cases.

"If you enable the inputs in the console you'll see that Studio One gives us nice fat input meters... These are large enough to be seen from the other side of the room where your microphones, guitars and other instruments probably are."

If you turn on the monitoring for the audio track the metering on that channel will switch to reflect the input monitoring. This will also follow through to the Main output where, if you've enabled K-14 metering, you can check that your input level is hitting that lowered zero point.

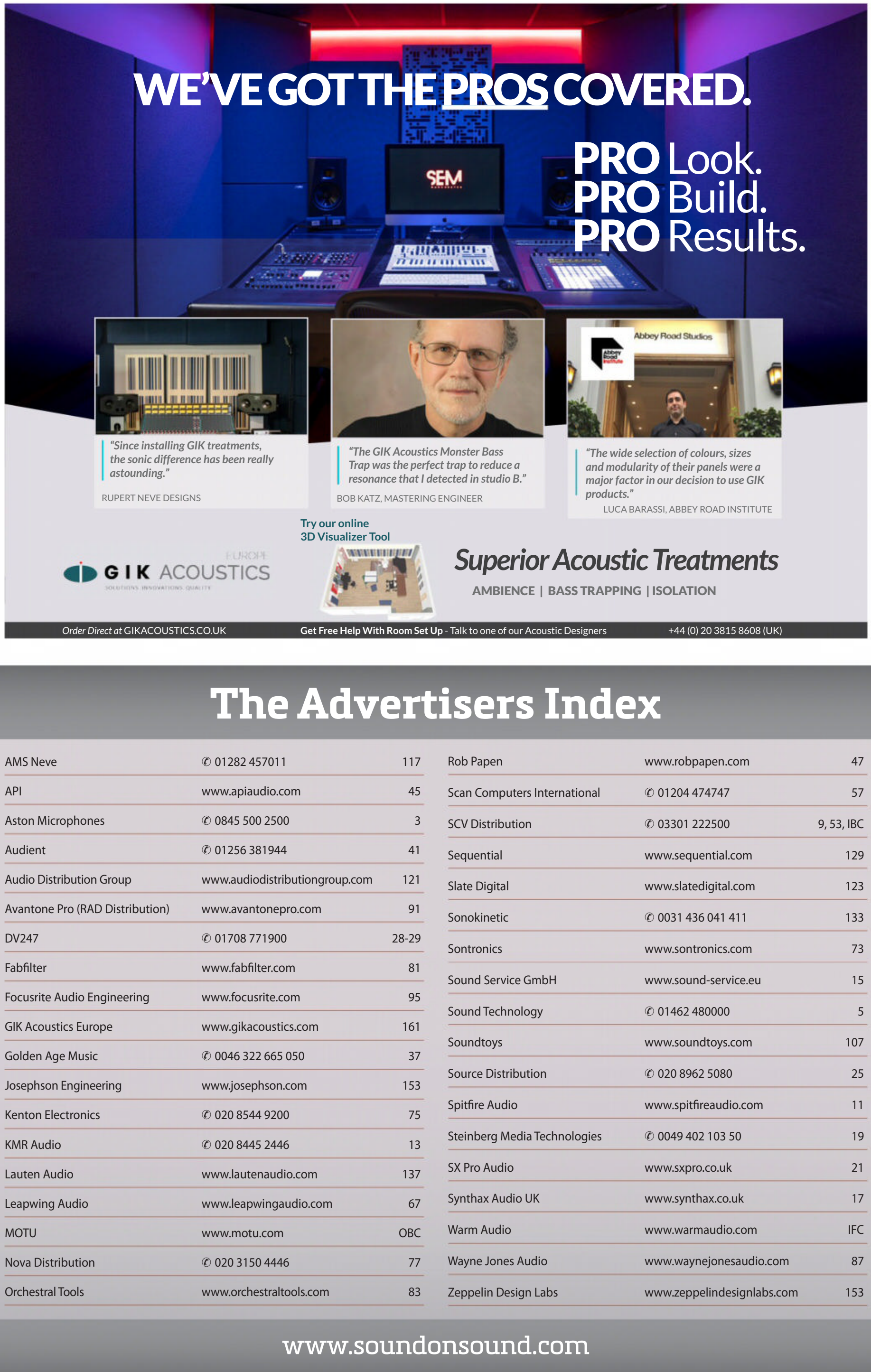
To help you with your quest for the perfect level, each input channel has a small gain knob at the top. This is able to add or subtract 24dB of gain to or from your signal, and you can also change the polarity here.

One more tip is that if you hover your mouse over the meter in any of the channels a white line appears at the mouse pointer tip along with the exact value of the meter at that point. Very handy for getting precise with your levels.

Metering Plug-ins

Amongst the PreSonus plug-ins you'll find a few useful little metering tools that give you even deeper investigations into your audio. These are more commonly used at the mastering stage, although the Level Meter in particular can be helpful for input monitoring and mixing. It gives you a lovely big visualisation of your levels on any channel you drop it onto. You can drag it about to resize it in either a horizontal or vertical orientation, and even have it fill your screen if you wish. It also includes the K metering system, which isn't available in regular channel metering, as well as an R128 EBU loudness option.

The Level Meter plug-in is a resizable metering window that you can drop into any channel in Studio One.



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SOS WHY I LOVE...

MY LAPTOP

ROBERT HICKS

In terms of music technology, it's been an exciting couple of decades. For a while now it has been possible to make broadcast-quality music straight from your laptop and this has liberated many from the confines of the windowless studio.

Studios remain completely necessary; after all, they're controlled spaces tuned to the needs of recording and audio engineering — but do you have to do everything creative there? Let's face it, most of us occupy that space in our home usually earmarked for storing cardboard boxes and offcuts of carpet. It made me wonder, do we instinctively associate the art of making music with traditional studio spaces with vast mixing consoles and rack gear with their LEDs idly glowing in the background? I suspect we probably do, and that association may dictate our workspace design.

The reality these days is that most of us build our studio around a computer. For me, the laptop became my principal choice as part of a strategy to migrate to a largely 'in-the-box' setup, believing I could perform a significant amount of my composing tasks from anywhere. Why escape the studio? Well, for me it's infinitely more inspiring to compose whilst sat in the garden. A view over the neighbouring fields, the sunshine enhancing vivid colours, bees going about their pollen collection, birds arriving and departing. A solid garden table, a fresh cup of coffee. It's a great place to be. Until it rains...

OK, it's true, if you're a professional creative, you still need a dedicated treated room with certain pieces of kit always on hand. But once you've recorded those guitars, why not edit and arrange them in the garden, on the train, or whilst you are sat next to the pool?

For me, it's important to regularly change the location of my working space. It keeps my ideas fresh and can often lead to clearer thinking and inspire alternative paths because, sometimes, having a dedicated room/studio can lead to a mental association with 'work' or 'isolation'.

With your laptop to hand, have a go at writing that woodwind passage or performing a rough mix on your headphones whilst the sun is shining. You might be surprised how inspiring it can be.

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